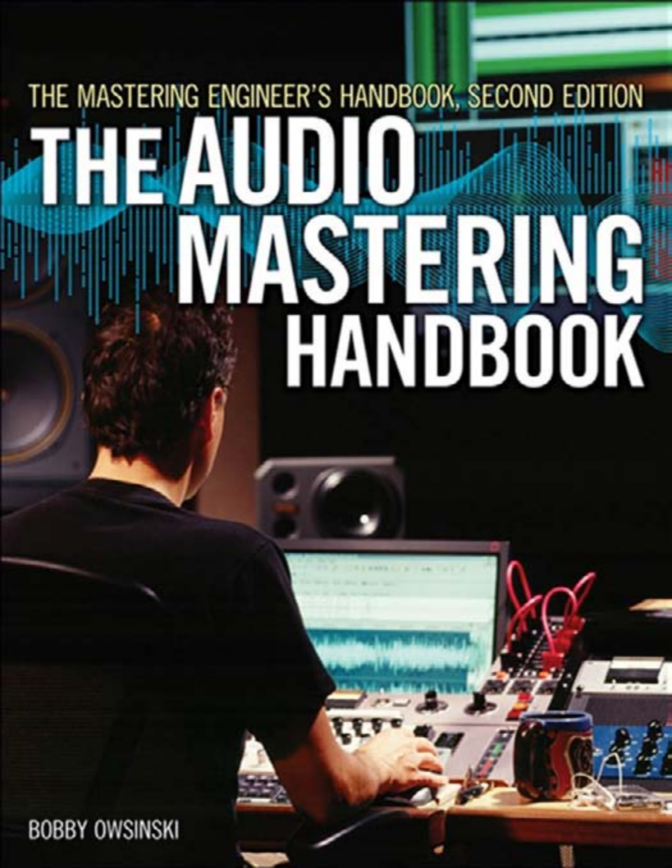
THE MASTERING ENGINEER'S HANDBOOK, SECOND EDITION

THE AUDIO MASTERING HANDBOOK



BOBBY OWSINSKI

THE Mastering Engineer's HANDBOOK

Second Edition: The Audio Mastering Handbook

by

Bobby Owsinski

THOMSON
COURSE TECHNOLOGY
Professional ■ Technical ■ Reference

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A special thanks to all the great mastering engineers who agreed to give a little insight into how they do what they do. It's great that you guys can be so open in such a competitive business. Also, special thanks to Mike Lawson, who turned me into a book author; Cathleen Small, for being the best editor I've ever had; Edward Colver, for taking the great cover pictures; John Zois from CDBits, for his great CD-pressing plant pictures; Clete Baker from Studio B, for his excellent vinyl photographs; and all the manufacturers that contributed graphics of their gear.

About the Author

A longtime veteran of the music industry, **Bobby Owsinski** has produced and composed for records, DVDs, motion pictures, and television shows. One of the first to delve into surround-sound music mixing, Bobby has worked on more than 200 surround projects and DVD productions for such diverse acts as Elvis, Jimi Hendrix, The Who, Willie Nelson, Neil Young, The Ramones, and Chicago, among many, many others.

Currently a principal in the music production house Surround Associates and content creator 2B Media, Bobby has also penned several hundred articles for many popular music and audio trade publications and has authored three books that are now staples in audio-recording programs in colleges around the world. A frequent moderator, panelist, and program director of a variety of music and professional audio industry conferences, Bobby has served as the longtime producer of the annual Surround Music Awards and is currently an executive producer for the *Guitar Universe* and *Desert Island Music* television programs. He is also a partner in the popular Asia Los Feliz restaurant in Los Angeles, and serves on the board of directors of the Media Entertainment Technology Alliance.

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Introduction

It's been eight years since the first version of *The Mastering Engineer's Handbook* came out and, boy, have things changed. It's safe to say that there has been a mighty revolution in the mastering world, with old technologies replaced and new ones continually evolving. Gone are the days of tape machines (for the most part), and soon even the CD might be a thing of the past. Gone (again, for the most part) are the days of "heavy iron" customized outboard gear that was necessary for a high-quality mastering job. Even though the basic mastering tools are still the same, they've mostly moved into the world of the DAW, so even someone with the most entry-level system now has access to powerful tools that only the top pros used to have access to. And maybe best of all, it's now possible to finish almost any kind of audio for any kind of distribution (which is what mastering really is) at home, in your small studio or bedroom.

But just because you can, doesn't mean that it's always a good idea to try to be the mastering engineer yourself. A lot of harm can come from misuse of the tools of mastering because the process and concepts are not really understood.

And that's what this book is about.

What we'll try to do is take a look at how the pros perform their magic, listen to them describe their processes in interviews, and develop a good, strong reference point where we can either do it ourselves and hopefully do no harm to the material (just like a doctor), or know when to call a pro and properly prep the program for them to get the best results possible.

More so than any other process in audio, mastering is more than just knowing the procedure and owning the equipment. Yes, more than any other job in audio, mastering done at its highest level is about the long, hard grind of experience. It's about the cumulative knowledge gained from 12-hour days of listening to both great and terrible mixes; from working on all types of music, not just the type you like; from saving the client's butt without him ever knowing it; from doing 10 times more work than the client ever sees.

Although I don't want to call myself a "mastering engineer" per se, since it's not a job I do every day, it's a process I know pretty well because

I've hung out in major mastering studios for many years (both as a client and socially), I have some very good friends who are world-class mastering engineers, and I have even taught some college courses on the subject.

So among the many things this book will provide is an insider's look at the process, not so much from my eyes, but from that of the legends, greats, and potential future-greats of the business.

My goal with this book is a simple one: To keep the guy who wants to do his own mastering out of trouble and help him do a better job, and to show that there's a lot more to a professional mastering job than meets the eye.

For those of you who have read my previous books, *The Mixing Engineer's Handbook, Second Edition* (Thomson Course Technology PTR, 2006) and *The Recording Engineer's Handbook* (ArtistPro, 2004), you'll notice that the format of this book is similar. It's divided into three sections:

- **Part I: The Mechanics of Mastering** gives an overview of the history, tools, philosophy, background, and tips and tricks used by the best mastering engineers in the business.
- **Part II: Audio Delivery Formats** provides some interesting and hard-to-find info on the delivery methods for the past and fading audio delivery formats—the vinyl record, CD and DVD, and the current audio delivery formats, such as MP3s, streaming audio, and high-definition discs (such as Blu-ray and HD-DVD).
- **Part III: The Interviews** provides a behind-the-scenes look at the mastering world through the eyes of some of the finest (and, in some cases, legendary) mastering engineers in the world.

Meet the Mastering Engineers

Here's a list of the engineers who contributed to this book, along with some of their credits. I've tried to include not only the most notable names in the business from the main media centers, but also engineers who deal with specialty clients. I'll be quoting them from time to time, so I wanted to introduce them early on so you have some idea of their background when they pop up.

- **Doug Sax.** Perhaps the godfather of all mastering engineers, Doug became the first independent by starting his famous Mastering Lab in Los Angeles in 1967. Since then, he has worked his magic with such diverse talents as The Who; Pink Floyd; The Rolling Stones; The Eagles;

Kenny Rogers; Barbra Streisand; Neil Diamond; Earth, Wind & Fire; Diana Krall; Dixie Chicks; Rod Stewart; Jackson Browne; and many, many more.

- **Bernie Grundman.** One of the most widely respected names in the recording industry, Bernie Grundman has mastered literally hundreds of platinum and gold albums, including some of the most successful landmark recordings of all time, such as Michael Jackson's *Thriller*, Steely Dan's *Aja*, and Carole King's *Tapestry*. A mainstay at A&M records for 15 years before starting his own facility (Bernie Grundman Mastering) in 1984, Bernie is certainly one of the most celebrated mastering engineers of our time.

- **Bob Ludwig.** After having worked on literally hundreds of platinum and gold records and mastered projects that have been nominated for scores of Grammys, Bob Ludwig certainly stands among the giants in the mastering business. After leaving New York City to open his own Gateway Mastering in Portland, Maine, in 1993, Bob has proved that you can still be in the center of the media without being in a media center.

- **Greg Calbi.** Greg started his career as a mastering engineer at the Record Plant New York in 1973 before moving over to Sterling Sound in 1976. After a brief stint at Masterdisk from 1994 to 1998, Greg returned to Sterling as an owner, where he remains today. Greg's credits are numerous, including Bob Dylan, John Lennon, U2, David Bowie, Paul Simon, Paul McCartney, Blues Traveler, and Sarah McLachlan, among many, many others.

- **Glenn Meadows.** Glenn is a two-time Grammy winner and a multi TEC award nominee who has worked on scores of gold and platinum records for a diverse array of artists, including Shania Twain, LeAnn Rimes, Randy Travis, Delbert McClinton, and Reba McEntire, as well as for multi-platinum producers such as Tony Brown, Jimmy Bowen, and Mutt Lange.

- **Eddy Schreyer.** Eddy opened Oasis Mastering in 1996 after mastering stints at Capitol, MCA, and Future Disc. With a list of chart-topping clients that span the various musical genres, such as Babyface, Eric Clapton, Christina Aguilera, Kanye West, Avenged Sevenfold, Fiona Apple, Hootie and the Blowfish, Offspring, Korn, Dave Hollister, Pennywise, Xzibit, Jesse Powell, and Tupac, Eddy's work is heard and respected worldwide.

- **Bob Olhsson.** After cutting his first number-one record (Stevie Wonder's *Uptight*) at age 18, Bob worked on an amazing 80 top-ten records while working for Motown in Detroit. Now located in Nashville, Bob's insightful account of the history of the industry makes for a truly fascinating read.

- **David Cheppa.** David began cutting vinyl in 1974 and since that time has cut almost 22,000 sides. He is the founder of Better Quality Sound, which is currently one of the few remaining mastering houses dedicated strictly to vinyl. Thanks to his intense interest and design engineering background, David has brought a medium once given up for dead to new, unsurpassed heights of quality.

- **Bob Katz.** Co-owner of Orlando, Florida–based Digital Domain, Bob specializes in mastering audiophile recordings of acoustic music, from folk music to classical. The former technical director of the widely acclaimed Chesky Records, Bob's recordings have received disc-of-the-month recognition in *Stereophile* and other magazines numerous times, and his recording of *Portraits of Cuba* by Paquito D'Rivera won the 1997 Grammy for Best Latin-Jazz Recording. Bob's mastering clients include major labels EMI, WEA-Latina, BMG, and Sony Classical, as well as numerous independent labels.

Part I

The Mechanics of Mastering

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CHAPTER 1

What Exactly Is Mastering?

Technically speaking, mastering is, quite simply, the intermediate step between taking the audio fresh from mixdown from a studio and preparing it to be replicated or distributed. But it is much more than that.

Mastering is the process of turning a collection of songs into a record by making them sound like they belong together in tone, volume, and timing (spacing between songs).

Mastering is not a set of tools or a device that music is run through and automatically comes out mastered (despite what the adverts for these types of so-called "mastering devices" say). It's an art form that, when done conscientiously in its highest form, mostly relies on an individual's skill, experience with various genres of music, and good taste.

BERNIE GRUNDMAN: *I think that mastering is a way of maximizing music to make it more effective for the listener as well as maybe maximizing it in a competitive way for the industry. It's the final creative step and the last chance to do any modifications that might take the song to the next level.*

GLENN MEADOWS: *I think that mastering is, and always has been, the real bridge between the pro audio industry and the hi-fi industry. We're the ones who have to take this stuff that sounds hopefully good or great on a big professional monitor system and make sure it also translates well to the home systems. We're the last link to get it right or the last chance to really screw it up and make it bad, and I think we're all guilty at times of doing both.*

Some History

In the early days of vinyl, mastering was a black art practiced by technical curmudgeons who mysteriously made the transfer from the electronic medium of magnetic audio tape to the physical medium of vinyl. There was a high degree of difficulty in this process because the level applied to the vinyl lacquer was so crucial. Too low a level and you get a noisy disk; hit it too hard and you destroy the disk and maybe the $15,000 (that's in 1950's and 1960's dollars) cutting stylus too.

Along the way, mastering (back then sometimes called *transfer*) engineers found ways to make the disks louder (and therefore less noisy) by applying equalization and compression. Producers and artists began to take notice that certain records would actually sound louder on the radio, and if they played louder, then the general public usually thought they sounded better, so maybe (they were speculating here) the disk sold better as a result. Hence, a new breed of mastering engineer was born, this one with some creative control and ability to influence the final sound of a record rather than just being a transfer jock from medium to medium.

Today's top mastering engineers practice less of the black art of disk cutting but no less the wizardry as they continue to subtly shape and mold the variations of frequencies and dynamics of a project. And that's the same goal if you're doing the mastering yourself.

From Vinyl to the CD and Beyond

Until 1948, there was no distinction between audio engineers because everything was recorded directly onto vinyl (all records were 10" and played at 78 RPM). In 1948, however, the age of the "transfer" engineer began when Ampex introduced its first commercial magnetic tape recorder. With most recording now being done to magnetic tape, a transfer had to be made to a vinyl master for delivery to the pressing plant; hence the first incarnation of the "mastering engineer" was born.

In 1955, Ampex released Sel-Sync (*Selective Synchronous*) recording, which gave the multitrack recorder the ability to overdub. Now that the recording industry was forever changed, so began the real distinction between the recording and mastering engineer, since the jobs now differed so greatly.

In 1957, the stereo vinyl record became commercially available and really pushed the industry to the sonic heights that it has reached today.

(Some say the best audio ever came from this era.) At this point the mastering engineer became more influential thanks to judicious and creative use of equalization and compression to cut the discs and make them sound better than when they were recorded.

With the introduction of the CD in 1982, the mastering engineer was forced into the digital age, but still used tools from the vinyl past. But with the 1989 introduction of the Sonic Solutions digital audio workstation with pre-mastering software, mastering gradually developed into its current digital state.

In the first half of 1995, MPEG-1 Audio Layer 3 files, more commonly referred to as *MP3s*, began to spread on the Internet, and their small file size set about a revolution in the music industry that continues to this day. This meant that the mastering engineer had to become well versed in how to get the most from this format (something it took years for many mastering engineers to get the hang of).

In 1999, 5.1 surround sound, high sample rates, and 24-bit word lengths took the mastering engineer into new, uncharted, but highly creative territory. By 2002, almost all mastering engineers had become well acquainted with the computer because virtually every project was edited and manipulated in a DAW.

Why Master Anyway?

Mastering should be considered the final step in the creative process because it is your last chance to polish and fix your project. This is the case in the United States, but in Europe mastering is looked upon as the first stage of the manufacturing process because it is the place where the digital bits get transferred to either a mechanical medium (such as vinyl) or another electronic medium better suited for mass production (such as CDs or cassettes). Both of these views are true, but it's a shame to overlook the creative aspect. It has become a moot point anyway, with many music releases completely bypassing CDs and the many other legacy media.

A project that has been mastered (especially at a top-flight mastering house) simply sounds better. It sounds complete, polished, and finished. The project that might have sounded like a demo before now sounds like a record. This is because the mastering engineer has added judicious amounts of EQ and compression to make the project bigger, fatter, richer, and louder. He has matched the levels of each song so they all have the same apparent level. He has fixed the fades so that they're smooth. He has edited out bad parts so well that you didn't even notice. He has made all

the songs blend together into a cohesive unit. In the case of mastering for CD, he has inserted the spreads (the time between each song) so the songs now flow together seamlessly. He has sequenced the songs so they fall in the correct order. He has proofed your master before it's sent to the replicator to make sure it's free of any glitches or noise. He has also made and stored a backup clone in case anything should happen to your cherished master, and he has taken care of all of the shipping to the desired duplication facility if you're using one. And all this happened so quickly and smoothly that you hardly knew it was happening.

Why It Sounds So Good When the Pros Do It

There are a lot of reasons why a commercial mastering facility usually produces a better product than when you master at home. First of all, the mastering house is better equipped. They have many things available that you probably won't find in a simple home or a small studio DAW room, such as high-quality digital transfer consoles, high-end A/D and D/A converters, ultra-smooth outboard compressors and equalizers, multiple tweaked 1/2" and 1/4" two-track tape machines (if needed), DAT machines (again, if needed), and an exceptional monitoring system.

The monitor systems of these facilities sometimes cost far more than many entire home studios. Cost here isn't the point, but quality is, since you can rarely hear what you need to hear on the commonly used nearfield monitors that most recording studios have in order to make the adjustments that you need to make. The vast majority of monitors and the rooms in which they reside are just not precise enough.

GLENN MEADOWS: *The reason people come to a mastering engineer is to gain that mastering engineer's anchor into what they hear and how they hear it and the ability to get that stuff sounding right to the outside world.*

EDDY SCHREYER: *You can't make a move or create a fix if you can't hear it, so obviously the mastering environment is extremely important. A great facility to me means both client services and a comfortable place that's able to facilitate both large and small sessions. I am assuming my studio is somewhat the norm. I can seat about five to six people in my room very comfortably, and I believe that is probably somewhat common. I think a mastering room that's too small is not a good thing. At times there are more than two or three people who want to show up at a mastering session, so that part of the client relationship is very important to me. So the facility sort of dictates what your goal is in terms of the client/engineer relationship and just how comfortable you want these people to be.*

Experience Is the Key

But the mastering engineer is the real key to the process. This is all he does day in and day out. He has "big ears" because he masters for at least eight hours every day and knows his monitors the way you know your favorite pair of sneakers. Plus, his reference point of what constitutes a good-sounding mix is finely honed thanks to working hours and hours on the best- and worst-sounding mixes of each genre of music.

GREG CALBI: *As far as the person who might be trying to learn how to do his own mastering, or understand mastering in general, the main thing is that all you need is one experience of hearing somebody else master something. Your one experience at having it sound so incredibly different makes you then realize just how intricate mastering can be and just how much you could add or subtract from a final mix.*

BERNIE GRUNDMAN: *Most people need a mastering engineer to bring a certain amount of objectivity to their mix, plus a certain amount of experience. If you (the mastering engineer) have been in the business a while, you've listened to a lot of material, and you've probably heard what really great recordings of any type of music sound like. So in your mind you immediately compare it to the best ones you've ever heard. You know, the ones that really got you excited and created the kind of effect that producers are looking for. If it doesn't meet that ideal, you try to manipulate the sound in such a way as to make it as exciting and effective a musical experience as you've ever had with that kind of music.*

DAVE COLLINS: *I personally think experience is as valuable as equipment in a large sense, because after you've done it for 10 or 20 years, you've heard almost everything that can possibly go wrong and go right on a mix. So you can, in one respect, quickly address people's problems.*

When a guy writes a book, he doesn't edit the book himself. He sends it off to an editor, and the editor reads it with a fresh set of eyes, just like a mastering engineer hears it with a fresh set of ears.

GLENN MEADOWS: *I don't mean to be arrogant, but it has to do with the experience of the engineer working in his environment. He's in the same room every day for years. I can walk into this room in the morning and know if my monitors are right or wrong just by listening to a track from yesterday. To me, that's the value of a mastering engineer. What they bring to the table is the cross-section of their experience and their ability to say, "No, you really don't want to do that."*

BOB OLHSSON: *To me it's a matter of trying to figure out what people were trying to do, and then doing what they would do if they had the listening situation and experience that I have.*

GLENN MEADOWS: *I find that the real value of a mainstream mastering facility versus trying to do it yourself or doing it in a small backwoods-type place or a basement place is that the experience of the engineer comes into play and it can save you money and time.*

Finally, if mastering was so easy, don't you think that every big-time engineer or producer (or record company, for that matter) would do it themselves? They don't, and mastering houses are busier than ever, which should tell you something.

DAVE COLLINS: *Every so often I'll have a client that I work with all the time, and his budget is gone by the time he's ready to master. And so he says, "Well, I'll go in the studio and I'll hook up a Massenburg EQ, and I'll do a little equalization, and I'll put a compressor of some type on the output of it." But he'll ultimately call back and say, "Well, I don't know what I'm doing here. I'm just making it sound worse."*

And that's kind of analogous to some guy trying to edit his own writing. It is the impartial ear that you get from your mastering engineer that is valuable. All this equipment and new technology that we've got is a great thing, but you're really asking for someone who has never heard the record before to hear it for the first time fresh.

BERNIE GRUNDMAN: *Mastering is more than just knowing how to manipulate the sound to get it to where somebody wants it to go. I think that a lot of it is this willingness to enter into another person's world, and get to know it and actually help that person express what he is trying to express, only better.*

Although all of this may seem as if I'm trying to discourage you from doing your own mastering, that's really not the case. In fact, what I'm trying to do is give you a reference point, and that reference point is how the pros operate and why they are so successful. From there you can determine whether you're better served by doing it yourself or using a pro.

But the reason that you're reading this book is because you want to learn about all the tricks, techniques, and nuances of a major mastering facility, right? Read on, and I'll show you the hows and whys of these operations in detail.

CHAPTER 2

Some Digital Audio Basics

Now is probably a good time for a brief review of some of the basics of digital audio. Although you may be familiar with the sample rate and word length already, there always seems to be a lot of questions about the differences between file formats, such as AIFF and WAV, so we'll try to take care of them straight away.

Sample Rate and Word Length

Sample rate and word length determine the quality of a digital audio signal. To understand the significance of sample rate and word length and how they affect quality, a brief discussion is in order. Remember, this is a *brief* discussion that will only give you the general concepts of digital audio. If you really want to get under the hood of digital audio, refer to a book such as *Principles of Digital Audio* by Ken Pohlmann.

The analog audio waveform is measured by an analog-to-digital converter (called an *A to D*, *ADC*, or *A/D* converter) in amplitude at discrete points in time, and this is called *sampling*. The more samples per second of the analog waveform that are taken, the better digital representation of the waveform that occurs, resulting in greater bandwidth for the signal. Audio on a CD has a sampling rate of 44,100 times a second (or 44.1 kHz), which, thanks to a law of digital audio called the Nyquist Theorem, yields a maximum audio bandwidth of about 22 kHz. A sampling rate of 96 kHz gives a better digital representation of the waveform because it uses more samples, and it yields a usable audio bandwidth of about 48 kHz. A 192-kHz sample rate yields a bandwidth of 96 kHz. Therefore, the higher the sampling rate, the better the representation of the signal and the greater the audio bandwidth—which means it sounds better!

A digital word is somewhat the same in that more is better. The more bits in a digital word, the better the dynamic range—which means it sounds better! Every bit means 6 dB of dynamic range. Therefore, 16 bits yields a maximum dynamic range of 96 dB, 20 bits equals 120 dB DR, and 24 bits a theoretical maximum of 144 dB DR.

From this you can see that a high-resolution 96-kHz/24-bit format (usually just abbreviated 96/24) is far closer to sonic realism than the current CD standard of 44.1/16, and 192/24 even more so. The higher the sample rate, the greater the bandwidth, and therefore the better the sound. The longer the word length (more bits), the greater the dynamic range, and therefore the better the sound.

What all this means is that a mixing engineer now has a choice of sonic resolutions to mix to that was never available before. For the highest fidelity, a stereo mix at 192/24 (and even higher in the future) can be chosen, although most people probably won't hear it at that resolution. But thanks to optical disc media such as DVD, Blu-ray, HD-DVD, and whatever else comes along, mixers are no longer tied to the old CD-quality standard of 44.1 kHz at 16 bits, which we'll cover in depth in Chapter 6, "Mastering for CD."

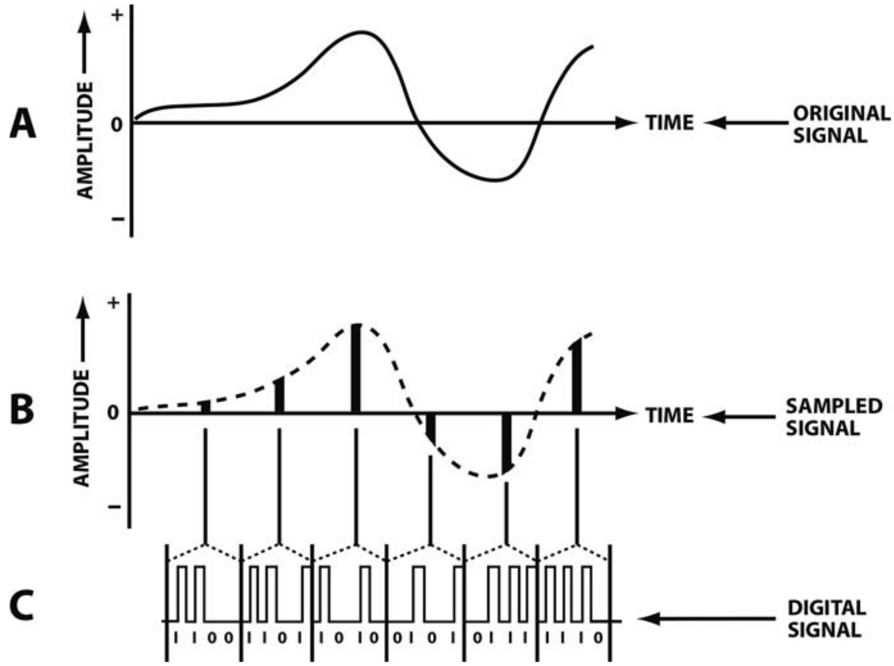
It's always best to mix to the highest resolution possible both for archival purposes and because a high-resolution master makes for a better sounding lower-resolution delivery. This applies even if the ultimate delivery medium is to be a lower resolution CD or MP3.

Standard Audio File Formats

This section discusses the types of files found on a typical digital audio workstation and their differences.

- **LPCM (*Linear Pulse Code Modulation*).** This is the process of sampling an analog waveform and converting it to digital bits that are represented by binary digits (ones and zeroes) of the sample values. When LPCM audio is transmitted, each one is represented by a positive voltage pulse and each zero is represented by the absence of a pulse (see Figure 2.1). LPCM is the most common method of storing and transmitting uncompressed digital audio. Because it is a generic format, it can be read by most audio applications, similar to the way a plain text file can be read by any word-processing program. LPCM is used by audio CDs and digital audio tape formats (DATs or DA-88s) and is represented in a file format on a DAW by AIFF, BWF, WAV, or SD2 files.

Figure 2.1
Linear PCM.



- **AIFF (*Audio Interchange File Format*).** This is a file format for storing LPCM digital audio data. It supports a variety of bit resolutions, sample rates, and channels of audio. The format was developed by Apple Computer and is the standard audio format for Macintosh computers, although all platforms can read almost any file format these days. AIFF files generally end with an .aif extension.

- **WAV (*Waveform Audio*).** This is another file format for storing LPCM digital audio data. Created by Microsoft and IBM, WAV was one of the first audio file types developed for the PC. WAV files are indicated by a .wav suffix in the file name and are often spelled wav (instead of wave) in writing. The WAV file format supports a variety of bit resolutions, sample rates, and channels of audio.

- **BWF (*Broadcast Wave*).** This is special version of the standard WAV audio file format developed by the European Broadcast Union in 1996. BWFs contain an extra "chunk" of data, known as the *broadcast extension chunk*, that contains information on the author, title, origination, date, time, and so on of the audio content. Perhaps the most significant aspect of BWFs is the feature of time stamping, which allows files to be moved from one DAW application to another and easily aligned to their proper point on a timeline or edit decision list. These files end with a .bwf file extension.

- **SDII or SD2 (*Sound Designer II*).** This is a mono or stereo audio file format for storing LPCM, originally developed by Digidesign for their DAW software applications. It is the successor to the original monophonic Sound Designer I audio file format. When used on a PC, the file must use the extension of .sd2. SD2 files are fast losing favor to the AIFF and WAV formats and should be considered obsolete.

Data Compression

Linear PCM files are large and, as a result, painfully slow to upload and download, even with a dedicated high-speed connection. As a result, data compression was introduced to keep a certain amount of sonic integrity (how much is in the ear of the beholder) while making an audio file imminently transportable.

Data compression isn't at all like the audio compression that we'll be talking about in the book. Data compression reduces the amount of physical storage space and memory required to store a sound, and therefore reduces the time required to transfer a file. We'll talk more about data-compressed files such as MP3, AC-3, Dolby Digital, DTS, and more in the delivery format chapters later in the book. (See Chapter 8, "Mastering for Internet Distribution," and Chapter 11, "Mastering for Film and Television.")

CHAPTER 3

Tools for Mastering

Someone once said that mastering is about 30 percent tools and 70 percent ears. That being said, the tools that are required are very unique to the genre, and in the analog days, they were often custom-made. Even today there are custom mastering versions of some very popular outboard recording units (again, mostly analog). These mastering versions have many of the most used controls detented and selectable, which is a rather expensive feature.

BERNIE GRUNDMAN: *We build our own equipment. It's built mostly as an integrated system to avoid a lot of extra electronics and isolation devices and so forth. We have all separate power to each one of our rooms and a very elaborate grounding setup, and we've proven to ourselves that it helps time and time again. We have all custom wire in the console. We build our own power supplies as well as everything else—the equalizers, everything.*

Common Elements

All tools for mastering, regardless of whether analog or digital, have two major features in common—extremely high sonic quality and repeatability. The sonic quality is a must in that any device in either the monitor or signal chain should have the least effect possible on the signal. The repeatability is important (although less so now than in the days of vinyl) in that the exact settings must be repeated in the event that a project must be redone (as in the case of additional parts or changes being called for weeks later). Although this feature isn't much of a problem in the digital domain because the settings can be memorized, many analog mastering devices are still used, so these hardware devices require special "mastering" versions that have 1 dB or less segment selections on the controls (see Figures 3.1 and 3.2). These additions add seriously to the cost of the device.

Figure 3.1
GML 9500 mastering equalizer. (Image courtesy of George Massenburg Labs.)

Figure 3.2
Avalon AD2077 mastering equalizer. (Image courtesy of Avalon Designs.)

THE SIGNAL PATH

Just as a reference point, most major mastering facilities have both analog and digital signal paths, since so many of the tools and source materials exist in both domains. That being said, the overall signal path is kept as short as possible, with any unneeded items removed so the signal remains unaffected.

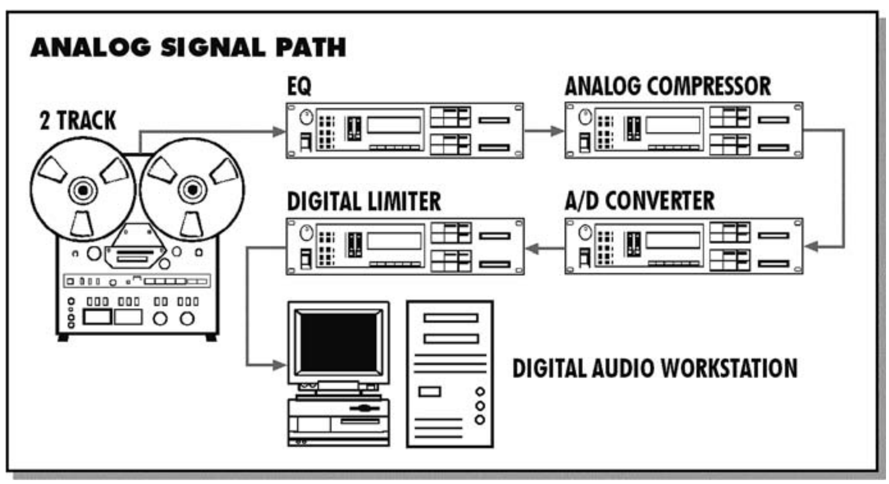


Figure 3.3
The analog signal path.

GREG CALBI: *On the analog side, what I try to do is combine light and dark, solid state and tube. So I have a bunch of tube equipment. I have the EAR compressors and the EAR EQs; the MEQ and the regular one, like the old Pultec. And I have an Avalon compressor and Avalon equalizer, which is a little bit more specific. I also have a Manley tube limiter compressor, one of those Vari-Mu's and one of Doug Sax's level amplifiers.*

DAVE COLLINS: *The analog signal path is a Studer 820 used just as a transport. We use a Flux Magnetics playback head that's connected to the outboard tape playback electronics…that is a half tube, half solid state. That feeds an all-custom analog console. Basically, the tape machine feeds some passive attenuation, and from there I've got a custom EQ that we use. I've got a Prism analog EQ, a Manley Variable-Mu compressor, and a heavily modified SSL console compressor, and we've got a Waves L2 limiter (serial number 0) and a dB Technology A/D converter. I also use that TC dB Max.…*

DOUG SAX: *As a point of interest, whether the source is analog or digital, if it needs EQ, I EQ it as an analog. That makes sense because if you come in with 96/24, I just look at it as good-sounding analog. I do what I want with it, then I'll get it down to 44.1 and 16 bit in the best way possible. So whether it's 1/2" or 1/4" analog or digital, it goes into good converters and comes up as analog. Then the EQ is passive with the same equalizer I've had since 1968. The limiters are all tubes and they're transformerless. Ninety-nine percent of what I do is done between those two devices.*

GLENN MEADOWS: *It can be a combination, but my path is typically 99 percent digital because 99 percent of what I am getting is digital. For example, with this one-inch two-track that I am working with, if I decide I need an analog EQ I will come through a Millenniia Dual (the mastering version with the detents on it), then run into my Prism AD2 converter, and then come into the rest of the mastering chain 24-bit digital. Then we will store it 24-bit digital and do anything else that we have to do at 24 bits internally. Then on the way back out the door, I can now loop out and back in and pick up my Z-Sys equalizer, using the power of POW-R word length reduction if I need to. The SADiE has the Apogee UV22 built in, if I decide to use that. So I have got the ability to handle it whichever way is most appropriate for the music. But the processing gear at the moment on the digital side is the Z-Systems six-channel EQ and Weiss EQ and compressor/limiters.*

BOB LUDWIG: *In the analog domain, it goes from the tape machine into George Massenburg/Sony electronics that are as minimal and audiophile as one can get. The output of that goes into either a dCS, Pacific Microsonics or sometimes Apogee analog-to-digital converter. When I need other outboard gear, we've got Neumann EQs and NTP and Manley compressors. Between the Manley, NTP, and digital domain compressors, that normally fills the bill for*

me, but I do have some Aphex Compellors. In the digital domain I have all the Weiss 96/24 stuff. The bw102, which has the 96-kHz de-esser in it as well, is complete with a mixer, compressor, and equalization.

As you can see, the analog path is somewhat of a hybrid in that it starts out in the analog domain but eventually enters the digital. Also, just because a source tape starts out in the digital domain (like a DAT), it doesn't necessarily mean that it will remain there. It's not uncommon for the mastering engineer to come back to analog in order to insert a specific equalizer or compressor, then return to digital.

THE DIGITAL DETANGLER

One of the few tools that seem to be universal among major mastering studios is one of the Z-Systems detanglers. This is essentially a digital router or patchbay that allows patching one digital device to another (or many others) at the push of a button. The unit functions as a digital audio patchbay, a distribution amplifier, a router, a format converter, and a channel switcher, all in one box (see Figure 3.4).

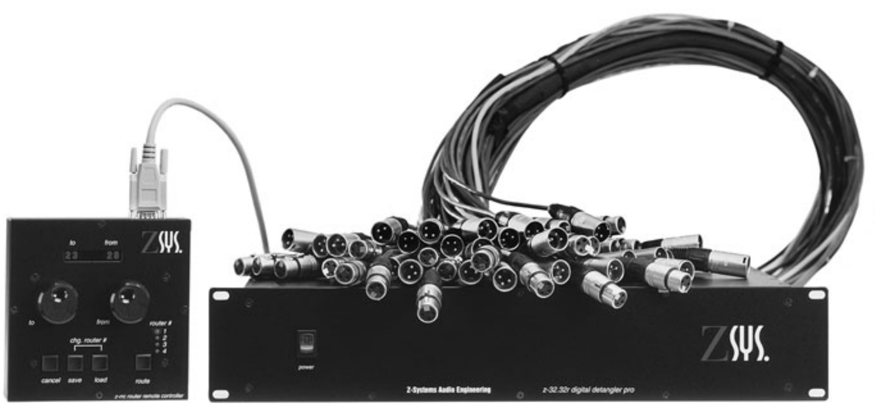


Figure 3.4
Z-Sys 32.32r digital detangler.
(Image courtesy of Z-Systems.)

For more information go to www.z-sys.com.

THE MONITOR SYSTEM

The heart and soul of the mastering signal chain are the chosen loudspeakers. More than any one device, these are the main link of the mastering engineer to both the reference point of the outside world and the possible deficiencies of the source material. More great pains go into the monitoring system than almost any other piece of gear in the studio.

BERNIE GRUNDMAN: *[P]robably the one biggest and most important piece of equipment that a mastering engineer can have is his monitor, and he has to understand that monitor and really know when it's where it should be. If you know the monitor and you've lived with it for a long time, then you're probably going to be able to make good recordings.*

THE ACOUSTIC ENVIRONMENT

Having the finest reproduction equipment is all for naught unless the acoustic environment in which it is placed is sound. Because of this, more time, attention, and expense are initially spent on the acoustic space than on virtually any other aspect.

BOB KATZ: *A great monitor in a bad room does absolutely nothing for you, so if you don't start with a terrific room and a plan for how it will integrate with the monitors, you can forget about it. No matter what you do, they will still suck, and you will still have problems.*

BOB LUDWIG: *To tell you the truth, I think a lot of people have heard about the effort we've gone through to make our room as acoustically perfect as possible. So many times people come into the room and they go, "Oh, my God!" or something like that. I felt that if I stayed in New York, I'd never be able to have a room that was acoustically as perfect as we knew how to make it. But in order to get as near perfect a situation as possible, you actually need a fairly large shell that's at least 30 feet long and accommodates a 17- or 18-foot ceiling.*

Because the room design is beyond the scope of this book, here's a list of some great acoustic designers for more information:

- Francis Manzella Design Limited: www.fmdesign.com
- Waterland Design: www.waterland.com
- Wave:Space, Inc.: www.wave-space.com
- Russ Berger Design Group: www.rbdg.com
- BOTO Design: www.BOTO.com
- Walters-Storyk Design Group: www.wsdg.com
- Bob Hodas Acoustic Analysis: www.bobhodas.com
- Chips Davis Designs: www.chips-davis.com
- Jeff Cooper Architects: www.jeffcooper.com
- TMH Corporation: www.tmhlabs.com
- Perception Incorporated: George Augsberger

MONITORS

The keys to a mastering monitor are wide and flat frequency response. Wide frequency response is especially important on the bottom end of the frequency spectrum, which means that a rather large monitor is required, perhaps with a subwoofer as well. This means that many of the common monitors used in recording and mixing, especially near-fields, will not provide the frequency response required for mastering.

Smooth frequency response is important for a number of reasons. First, an inaccurate response will result in inaccurate equalization in order to compensate. It will also probably mean you'll overuse the EQ as well in an unconscious attempt to overcome the deficiencies of the monitors themselves.

Large monitors with a lot of power behind them are not for loud playback, but for clean and detailed, distortion-free level. These monitors never sound loud; they just get bigger and bigger sounding and yet reveal every nuance of the music.

Although the selection of monitoring is a very subjective and personal issue (just as in recording), there are some brand names that repeatedly pop up in major mastering houses. These include Tannoy, B&W, Lipinski, and Duntech (see Figures 3.5 and 3.6).

Figure 3.5
B&W 801D. (Image courtesy of B&W Loudspeakers.)

Figure 3.6
Lipinski L-717. (Image courtesy of Lipinski Sound.)

- Tannoy: www.tannoy.com
- B&W: http://www.bowers-wilkins.com
- Duntech: www.duntech.com.au
- Lipinski Sound: www.lipinskisound.com

BOB LUDWIG: *One reason I've always tried to get the very best speaker I can is I've found that when something sounds really right on an accurate speaker, it tends to sound right on a wide variety of speakers. I've never been a big fan of trying to get things to sound right only on an NS-10M.*

EDDY SCHREYER: *I've been using Tannoys since about 1984 or 1985. I'm just a big fan of the dual-concentrics. I think the phase coherency is just unsurpassed. Once you get used to listening to these boxes, it's very difficult to listen to spread drivers again. In this particular case, my Dual 15s have been custom-modified for the room to some degree, and using them is just a great treat. I think they are one of the easier speakers to listen to since they certainly don't sound like the big brash monitor that they possibly might look to be. A typical comment made about the monitors here at Oasis is that they sound like the best big stereo system they've ever heard, which is a terrifically flattering compliment. I also have some little Tannoy System 600s for near-fields, and now I've added some dual 15 subs to the mains.*

BERNIE GRUNDMAN: *We build our own boxes and crossovers and we use all Tannoy components. We have it all mixed in with different elements that we feel are going to give us the best sound. It's not that we're going for the biggest or the most powerful sound; we're going for neutral because we really want to hear how one tune compares to the other in an album. We want to hear what we're doing when we add just a half dB at 5k or 10k. A lot of speakers nowadays have a lot of coloration, and they're kind of fun to listen to, but boy, it's hard to hear those subtle little differences. We just use a two-way speaker system with just one woofer and one tweeter so it really puts us in between near-fields and big soffited monitors.*

ON THE BOTTOM

Getting a project to have enough low end so that it translates well to speaker systems of all sizes is one thing that mastering engineers pride themselves on, and one of the reasons that near-field or even popular soffit-mounted large monitors are inadequate for mastering. The only way that you can properly tune the low end of a track is if you can hear it; therefore, a monitor with a frequency response to at least 40 Hz is definitely required.

SUBWOOFERS

To hear that last octave on the bottom, many mastering engineers are now resorting to subwoofers. A great debate rages as to whether a single subwoofer or stereo subwoofers are required for this purpose. Those who say stereo subs are a must insist that enough directional response occurs at lower frequencies to require a stereo pair. There is also a sense of envelopment that better approximates the realism of a live event with stereo subs. Either way, the placement of the subwoofers is of vital importance due to the standing waves of the control room at low frequencies.

For Best Subwoofer Placement

Place the subwoofer in the engineer's listening position behind the console.

Feed pink noise only into the subwoofer at the desired reference level. (Eighty-five dB SPL should do it, but the level isn't critical.)

Walk around the room near your main monitor speakers until you find the spot where the bass is the loudest. That's the spot to place the sub. For more level, move it toward the back wall or corner, but be careful because this could provide a peak at only one frequency. You're looking for the smoothest response possible (which may not be possible without the aid of a qualified acoustic consultant.)

Single Subwoofer Placement and Adjustment Tips

Though there is a totally scientific way to place the subwoofer, it is beyond the means of all but the largest facilities. Fortunately, there's a method that will get you in the ballpark, although you'll have to tweak a bit by experimenting from there. Keep in mind that this method is for single subwoofer use.

To Calibrate the Subwoofer

Using only one main speaker, feed pink noise in at a desired level (say 80 dB, for example) with the subwoofer disconnected.

Listening only to the subwoofer, set its level 6 dB less than the main speaker (74 dB). This applies if you're using an SPL meter, such as a Radio Shack. If you're using a real-time analyzer (RTA), the level of each band would be the same as your reference level (80 dB, in this case).

Adjust the phase of the subwoofer to the position with the most bass. This can be done by adjusting the phase control on the unit or by simply reversing the wires on the input connector.

Adjust the crossover point until the transition between the subwoofer and satellite is the most seamless.

AMPLIFIERS

Although the trend for most recording-style monitors is toward self-powered units, most speakers in the mastering environment still require an outboard amplifier—and a rather large one at that. It is not uncommon to see amplifiers of well over 1,000 watts per channel in a mastering situation. This is not for level (since most mastering engineers don't listen all that loudly), but more for headroom so that the peaks of the music induce nary a hint of distortion. Because many speakers used in a mastering situation are rather inefficient as well, this extra amount of power can compensate for the difference.

Although many power amps that are standard in professional recording, such as Manley, Bryston, and Hafler, are frequently used, it's not uncommon to see audiophile units such as Cello, Threshold, Krell, and Chevin (see Figure 3.7).

- Bryston: www.bryston.ca
- Chevin Research: www.chevin-research.com
- Threshold-Audio: www.threshold-audio.com
- Krell: www.krellonline.com
- Manley: www.manleylabs.com

Figure 3.7
Krell 302. (Image courtesy of Krell Industries.)

BOB LUDWIG: *When I started Gateway, I got another pair of Duntech Sovereigns and a new pair of Cello Performance Mark II amplifiers this time. These are the amps that will put out like 6,000-watt peaks. One never listens that loudly, but when you listen, it sounds as though there's an unlimited source of power attached to the speakers. You're never straining the amp, ever.*

CONVERTERS

With the advent of the digital age, mastering studios have been forced to add a new set of tools to their arsenal—analog-to-digital (A/D) and digital-to-analog (D/A) converters. Because each brand has a slightly different sound (just like most other pieces of gear), most mastering facilities have numerous versions of each type available for a particular type of music.

Among the current popular converters are Prism Sound, Lavry Engineering, Mytek, Apogee, and Benchmark Media (see Figure 3.8).

Figure 3.8
Lavry AD122 analog-to-digital converter. (Image courtesy of Lavry Engineering.)

GREG CALBI: *I usually work with two different A-to-D converters. I have a dB Technologies converter and I have one that the guys at JVC were fooling around with for awhile, which is excellent. I try to have two different converters at all times, one that maybe has a deeper bottom and better imaging, and another one that's maybe a little more exciting in the midrange.*

EQUALIZERS

One of the bread-and-butter tools of the mastering engineer, the equalizer—or, more accurately, a set of equalizers—is used more than almost any other device with the exception of the compressor. Mastering equalizers differ from their recording counterparts in that they usually feature stepped rather than continuously variable controls in order to be able to repeat the settings. The steps may be in increments as little as 0.5 dB, although 1 dB is seen most.

Popular analog hardware equalizers include the GML 8200 and 9500, the Avalon 2077, the Sontec MFS 432, and the Manley Massive Passive (see Figure 3.9). Some of the more popular digital hardware equalizers are the Weiss EQ-1 (see Figure 3.10) and the Z-Sys Z-Q1.

Popular software equalizers include the Sonnex Oxford EQ-500 (see Figure 3.11) and the Massenburg DesignWorks mdweq-v2 (see Figure 3.12).

Figure 3.9
Manely Massive Passive equalizer.
(Image courtesy of Manley Labs.)

Figure 3.10
Weiss EQ-1 digital equalizer.
(Image courtesy of Weiss.)

Figure 3.11
Sonnex Oxford EQ-500 plug-in. (Image courtesy of Sonnex.)

Figure 3.12
Massenburg Designworks mdweq-v2 plug-in. (Image courtesy of Massenburg DesignWorks.)

COMPRESSORS AND LIMITERS

The other major bread-and-butter tools of the master engineer are the compressor and the limiter. Although during recording this is usually the same unit that can be selected to function either way, mastering requires two separate units. Generally speaking, the compressor is used to shape the dynamics of a song by adding punch and strength, whereas the limiter is used to raise the apparent level of the song by controlling the musical peaks.

Hardware compressors that are often found in major mastering facilities include the analog Manley Vari-Mu (see Figure 3.13) and the Tube-Tech LCA 2B as well as the digital Junger d01, the Waves L2 (see Figure 3.14), and the TC M5000.

Some of the popular software compressors and limiters include the Oxford Dynamics 500w and the Waves L1 Ultramaximizer (see Figures 3.15 and 3.16).

Figure 3.13
Manley Vari-Mu compressor. (Image courtesy of Manley Labs.)

Figure 3.14
Waves L2 limiter. (Image courtesy of Waves Audio Ltd.)

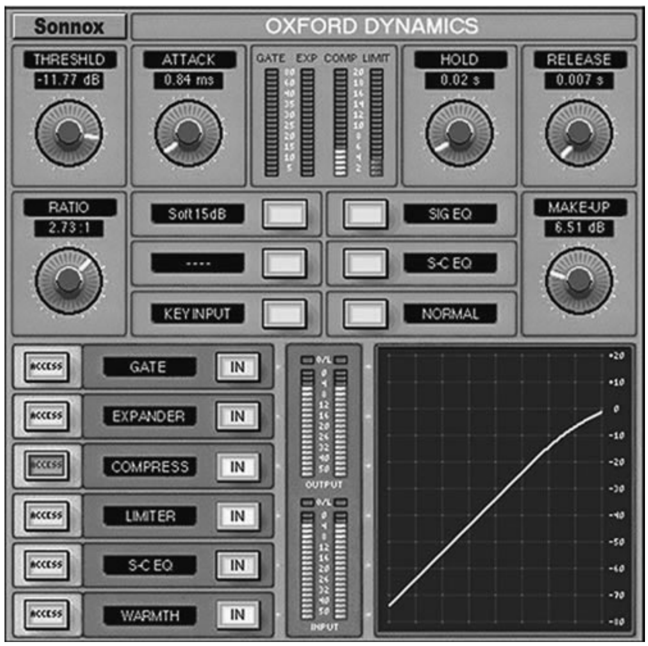


Figure 3.15
Oxford Dynamics 500w plug-in. (Image courtesy of Sonnex.)

Figure 3.16
Waves L1 Ultramaximizer plug-in. (Image courtesy of Waves Audio Ltd.)

TAPE MACHINES

Although the use of magnetic audio tape, either analog or digital, has decreased to minimal, it's still used enough that major mastering facilities must have the machines on hand.

Analog Tape Machines

Although the use of analog tape is very limited these days, you still see it used occasionally for the final mix, particularly on big-budget superstar sessions. Far and away the workhorse of the analog world is the 1/2" two-track tape machine (meaning it uses 1/2" magnetic tape), although this usually has a 1/4" headstack available as well. The most widely sought after machine for this purpose is the Ampex ATR-102 (see Figure 3.17), although many facilities have Studer 827s as well. It is not uncommon for the electronics of these machines to be highly modified to improve the signal path. It should be noted that neither machine is currently in production, meaning that they draw premium prices on the used market.

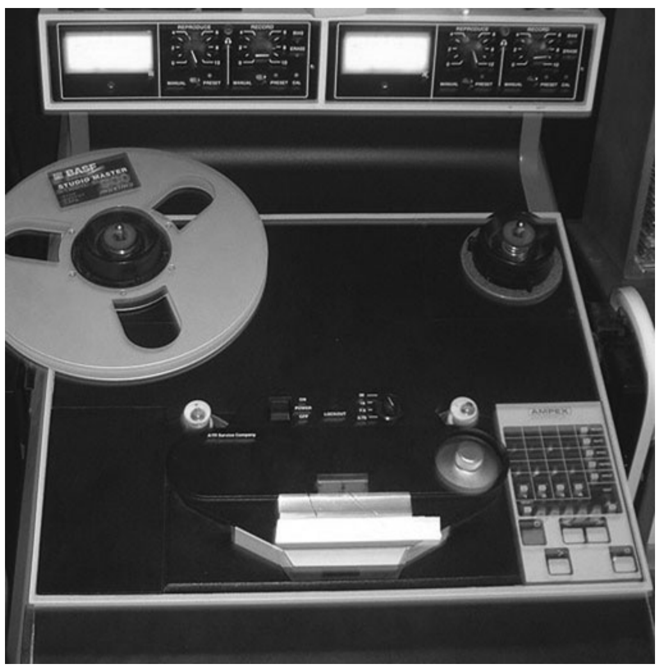


Figure 3.17 Ampex ATR-102 two-track recorder.

When analog tape was at its peak in the mid-1990s, a format that was briefly used was the 1" two-track. Again, this is a 1" headstack mounted on an Ampex or Studer transport.

At one point in time, cassette decks were an important part of the mastering facility, with huge banks of decks used for artist and label check copies. But since the advent of the inexpensive CD burner, cassettes have nearly gone the way of the dinosaur. However, mastering facilities usually have one around just in case it's needed. Most major facilities haven't even powered their decks on in years, though.

BOB KATZ: *My analog path starts with a custom-built set of Ampex MR70 electronics, which in my opinion are the best playback electronics that Ampex ever invented. I have that connected to a Studer C37 classic 1964 vintage transport with the extended low-frequency heads that John French put in, made by Flux Magnetics. It's just real transparent and not tubey-sounding at all, just open and clean.*

BOB LUDWIG: *We've got six different ways of playing back analog tape. We've got a stock Studer A820. We've got a Studer that's got Cello class A audiophile electronics. We've got a stock ATR, a tube ATR, and an unbalanced ATR. We also have one of the Tim de Paravicini 1" two-track machines with his fantastic tube electronics. When you record with his custom EQ curve at 15 ips, it's basically flat from eight cycles up to 28 kHz.*

GREG CALBI: *I have an ATR analog deck with tube electronics and one with solid state electronics. I also have a Studer 820. Most of the time at the beginning of an analog session, I'll play it off each of those three machines and see which one sounds the best. I usually work with two different A-to-D converters. I have a dB Technologies converter, and I have one that the guys at JVC were fooling around with for awhile, which is excellent. I try to have two different converters at all times, one that maybe has a deeper bottom and better imaging, and another one that's maybe a little more exciting in the midrange.*

DIGITAL TAPE MACHINES

Although any machine pulling magnetic tape with digital bits stored on it is now obsolete, it's still a good idea to know a little about them. You never know when the info might come in handy.

DAT

Although for a brief few years the DAT machine was the delivery king to the mastering studio, it's now become another rarely used tape format. If and when it's used, the A/D and D/A converters are usually bypassed for ones of higher quality. The limiting factor of the typical DAT is that it's a 16-bit medium, although a 24-bit Tascam DA-45HR was later introduced to overcome that limitation.

Sony PCM-1630

When digital audio mastering first began, one of the staples of the mastering scene was the Sony PCM-1630 (see Figure 3.18), which is a digital processor connected to either a Sony DMR 4000 or BVU-800 3/4" U-matic video (yes, video) machine. Since the beginning of the CD, the 1630 was the standard format that the mastering facility used to deliver the master to the replicator because of its low error rate. Although every facility at one time had a least one, and they once drew premium prices on the used market, the 1630 has long been obsolete. Many major mastering facilities still have one around, though, just in case they have to retrieve a master in this format.

Figure 3.18
Sony PCM-1630.

CONSOLES

Although mastering consoles (sometimes referred to as *transfer* consoles) at one time were much more sophisticated and were the centerpiece of the master studio, these days mastering consoles can be as simple as a piece of wire with relays in the middle to connect the various pieces of gear and control the monitor level. A mastering console differs from a normal recording console in that there are only two inputs for stereo (four at most for manual crossfades between songs) and no channel or track assignments. And because most of the processing like EQ and compression/limiting comes from specialized outboard devices, the console can be the virtual "straight wire with gain."

Due to the unique nature and relatively small size of the mastering market, few companies currently manufacture dedicated mastering consoles. Manley Labs designs custom-built analog-based consoles, while Weiss (with their now-standard 102 modules), Crookwood, and SPL Labs manufacture console modules for the digital domain.

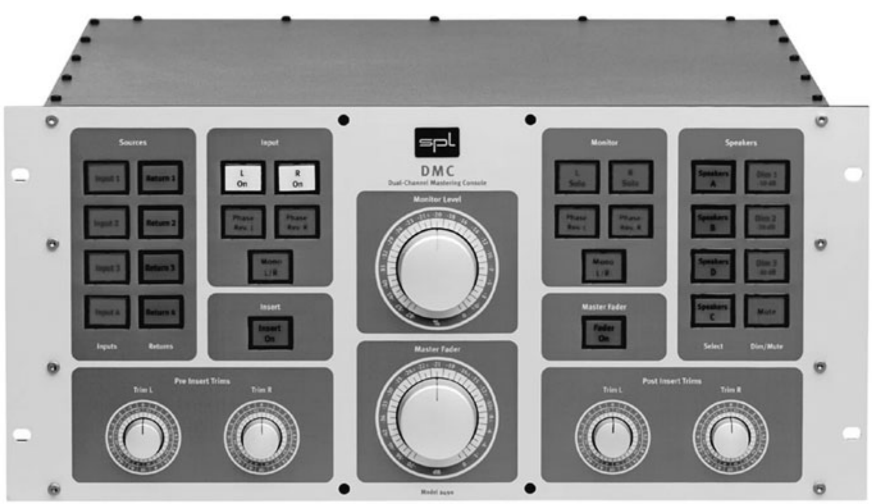


Figure 3.19
SPL DMC 960 mastering console. (Image courtesy of Sound Performance Labs.)

Figure 3.20
Crookwood monitoring panel.
(Image courtesy of Crookwood.)

THE DIGITAL AUDIO WORKSTATION

Although it is not always the case, the digital audio workstation (DAW) has now become the heart and soul of the mastering studio, allowing the engineer to complete tasks such as editing and sequencing with far greater ease than was ever thought possible. Plus, the DAW allows for new tasks to be carried out in ways that couldn't even be conceived of only 10 years ago.

The Big DAW Players

Although in a pinch almost any DAW can be used for mastering, a few manufacturers have established themselves as the mastering engineer's favorite, primarily because dedicated mastering features are included.

Sonic Studio

The PreMaster CD2 software is the latest offering from what used to be the premier mastering DAW manufacturer—Sonic Studio. Originally the audio division of Sonic Solutions (the company with the DVD authoring software), Sonic Studio was spun off into its own company in 2004. Originators of so much of what is now commonplace in DAWs (waveform display, 24-bit I/O, four-point editing, and premastered CD format, among other things), a Sonic system used to be a complete turnkey system built around an Apple Macintosh computer that included all of the I/O, DSP cards, and software needed to complete the system. With an abundance of excellent audio interfaces to choose from nowadays, as well as computers with adequate horsepower that no longer require the DSP card that used to be required, a Sonic system now consists of only a reasonably priced piece of software rather than a very expensive ($30,000 or so) turnkey system.

Cube-Tec

Designed specifically for the mastering market, the German Cube-Tec AudioCube is an up to 192-kHz/24-bit integrated Windows/PC-based complete turnkey system. Based around Steinberg's WaveLab software, AudioCube's major software feature is powerful plug-ins called *Virtual Precision Instruments*, or *VPIs*. A series of VPIs are specifically designed for mastering, restoration, and signal analysis, which is why the AudioCube has gained favor with mastering engineers worldwide recently.

SADiE

Another turnkey system is the British SADiE. With many features built for mastering (dithering, speed and pitch adjustment, creation and positioning of PQ points, UPC and ISRC code insertion, DDP disk image creation), the SADiE is both fast and easy to use.

Other popular DAWs, such as Pro Tools, Digital Performer, Sound Forge, and the like, while very good editors, lack the necessary tools that a mastering engineer routinely uses, such as ISRC insertion, PQ code editing, and elegant and powerful fade options.

OTHER DEVICES

There are a few common mastering devices that are widely used and very important in day-to-day mastering.

Metering

Precise and accurate metering are essential for the mastering engineer, so in many cases an outboard device is added. Although the modern mastering studio is loaded with peak-reading digital meters, most mastering engineers still like to use a good old-fashioned VU meter as well. This is because the VU gives a more accurate indication of the relative loudness than a peak meter. The classic example of this is the human voice, where a very quiet voice can have an extremely high peak level. It "looks" loud on a digital meter, but it sounds quiet. Because of its mechanical properties and ballistics, a VU meter "looks" at the signal closer to the way we hear than a peak meter does.

A VU meter doesn't have nearly the precision necessary for modern mastering, however, because the mastering engineer is constantly concerned about peaks and digital overs. That's why most mastering facilities use precision metering from manufacturers such as Dorrough (see Figure 3.21), Mytek, Logitek, and RTW.

Figure 3.21
Dorrough 40-A loudness meter. (Image courtesy of Dorrough Electronics, Inc.)

Sample Rate Converters

It is not uncommon for a DAT to be delivered at a sample rate other than the standard 44.1 kHz used for CD, and therefore a sample rate converter (SRC) is sometimes necessary. Although this function is sometimes available within the DAW, this is a complicated DSP task requiring massive calculations that tends to change the sound. Therefore, most mastering engineers prefer to use a dedicated system for this task. Popular models include the Z-Systems 2-src and the Weiss SFC2 (see Figure 3.22).

Figure 3.22
Weiss SFC2. (Image courtesy of Weiss Electronics.)

De-Essers

One of the most important tools to the mastering engineer is the de-esser (see Figure 3.23). As the name implies, a de-esser limits the amount of *S* sounds that might occur in a vocal track. Excessive high-frequency content is sometimes a by-product of compression and is known as *sibilance*. A de-esser is a frequency-dependent compressor that only triggers when excessive selective frequency content is present. Although sibilance control is a somewhat greater concern when cutting vinyl (see Chapter 7, "Mastering for Vinyl"), it's still of utmost importance to the mastering engineer because sibilance can have a very negative effect on the quality of the program.

Figure 3.23
Weiss DS1 de-esser. (Image courtesy of Weiss.)

CHAPTER 4

The Mechanics of Mastering

The actual mechanics of mastering can be broken down into a few functions, namely maximizing the level of the various program elements; maintaining the frequency balance; and using the main functions of the DAW, such as editing, fades, and spreads, and PQ and ISRC insertion. What really separates the upper-echelon mastering engineers from the rest is the ability to make the music (any kind of music) as big and loud and tonally balanced as possible, *but with the taste to know how far to take those operations.* The DAW functions, on the other hand, are somewhat mechanical, and although there are tricks involved, they usually don't get the same amount of attention as the former.

Level

The amount of perceived audio volume, or level, without distortion (on an audio file, CD, vinyl record, or any other audio delivery method yet to be created) is one of the things on which many top mastering engineers pride themselves. Notice the qualifying words *without distortion*, since that is indeed the trick—to the make the music as loud as possible (and thereby competitive with other products) while still sounding natural. Be aware that this generally applies to modern pop/rock/R&B/urban genres and not as often to classical or jazz, whose listeners much prefer a wider dynamic range in which maximum level is not a factor.

COMPETITIVE LEVEL

The volume/level wars really began way back in the vinyl era of the 1950s, when it was discovered that if a record played louder than the others on the radio, the listeners would perceive it to be better-sounding, therefore

making it a hit. Since then it has been the charge of mastering engineers to make any song intended for radio as loud as possible in whatever way they can.

And of course, this applies to situations other than the radio as well. Take for instance the iPod, the CD changer, or, in the very old days, the record jukebox. Most artists, producers, and labels don't want one of their releases to play softer than their competitors' releases because of the perception (not necessarily the truth) that it wouldn't sound as good if it's not as loud.

But the limitation of how loud a "record" (we'll use this term generically) can actually sound is determined by the delivery medium to the consumer. In the days of vinyl records, if a mix was too loud the stylus would vibrate so much that it would lift right out of the grooves, and the record would skip. When mixing too hot to analog tape, the sound would begin to softly distort, and the high frequencies would disappear (although many engineers and artists actually like this effect). When digital audio and CDs came along, any attempt to mix beyond 0 dB Full Scale resulted in terrible distortion as a result of digital "overs." (Nobody likes this effect.)

So trying to squeeze every ounce of level out of the track is a lot harder than it seems, and that's where the art of mastering comes in.

HYPERCOMPRESSION: DON'T GO THERE!

That being said, over the years it has become easier and easier to get a record that's hotter and hotter in perceived level, mostly because of new digital technology that has resulted in better and better limiters. Today's digital "look ahead" limiters make it easy to set a maximum level (usually at –.1 or –.2 dB FS) and never worry about digital overs and distortion again, but this usually comes at a great cost in audio quality.

Too much buss compression or over-limiting, either when mixing or mastering, results in what's become known as *hypercompression*. Hypercompression is to be avoided at all costs because:

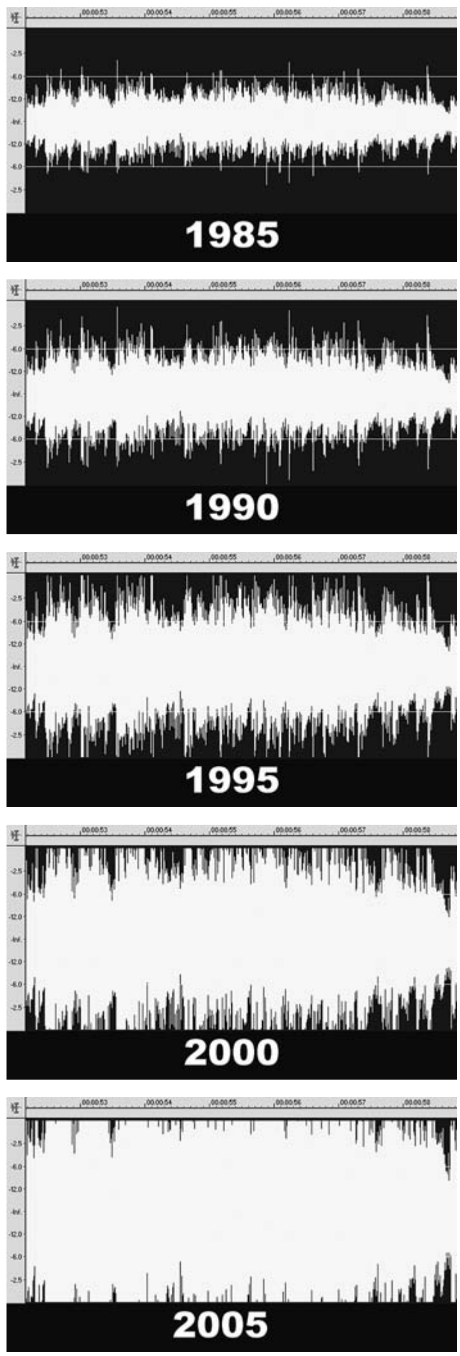
- It can't be undone later.
- It can suck the life out of a song, making it weaker-sounding instead of punchier.
- Lossy codecs (see Chapter 12, "Internet Delivery Formats") such as MP3 have a hard time encoding hypercompressed material and insert unwanted side effects as a result.
- It leaves the mastering engineer with no room to work.

- It's known to cause listener fatigue, so the consumer won't listen to your record for as long or as many times.

- A hypercompressed track can actually sound worse over the radio because of the behavior of broadcast processors at the station.

A hypercompressed track has no dynamics, leaving it loud but lifeless and unexciting. On a DAW, it's a constant waveform that fills up the DAW region. Figure 4.1 shows how the levels have changed on recordings over the years.

Figure 4.1
From very little compression to hypercompression.



This practice has come under fire as of late since we've just about hit the loudness limit, thanks to the digital environment that we now use. Still, both mixing and mastering engineers try to cram more and more level onto the disc, only to find that they end up with either a distorted or an over-compressed product. (Go back and listen to the Red Hot Chili Peppers' 1999 release *Californification* for a most egregious example.) Although this might be the sound that the producer/artist is looking for, it does violate the mastering engineer's unwritten code of keeping things as natural-sounding and unaltered as possible while performing his level magic.

EDDY SCHREYER: *What I am hearing is that various [mastering] houses are really over-compressing, trying to get more apparent level. The tradeoff with excessive compression to me is the blurring of not only the stereo image, but blurring the highs too. An over-compressed program sounds pretty muddy to me. In the quest to get the level, they end up EQing the heck out of these tracks, which of course induces even more distortion between the EQ and the compression.*

BOB LUDWIG: *When digital first came out, people knew that every time the light went into the red that you were clipping, and that hasn't changed. We're all afraid of the "over" levels, so people started inventing these digital domain compressors where you could just start cranking the level up. I always tell people, "Thank God these things weren't invented when the Beatles were around, because for sure they would've put it on their music and would've destroyed its longevity." I'm totally convinced that over-compression destroys the longevity of a piece. Now when someone's insisting on hot levels where it's not really appropriate, I find I can barely make it through the mastering session. I suppose that's well and good when it's a single for radio, but when you give that treatment to an entire album's worth of material, it's just exhausting. It's a very unnatural situation. Never in the history of mankind has man listened to such compressed music as we listen to now.*

BERNIE GRUNDMAN: *That's one of the unfortunate things about the industry, and it was even that way with vinyl. What happens is everybody is right at that ceiling level as high as you can go, so now guys without a lot of experience try to make things loud, and the stuff starts to sound god-awful. It's all smashed and smeared and distorted and pumping. You can hear some pretty bad CDs out there.*

BOB OLHSSON: *We can do things beyond anything we were ever able to do before, like turn the signal into a square wave even. The other thing is that people are commonly going too far with compression during mixing, so much that an awful lot of mixes can't be helped. I average a couple of mastering jobs a year where I can't do anything to it. If you switch anything in at all, it just absolutely turns to dust. All you can do is hope that the stations that play it won't destroy it too much more.*

DAVE COLLINS: *I never would've thought that we would be cutting CDs at this level. It's to the point where a large amount of our day is optimizing the gain structure in the console and checking what kind of limiter you're going to use and how you're going to use it just to get the CD as loud as you possibly can. I don't get it. I have to play the game because if you want to stay in business, you've got to compete on an absolute level, but it's really a horrible trend. I wish all mastering engineers would speak out about this because it sucks.*

I buy CDs that I really want to listen to, and they are so fatiguing. It's impossible to get that amount of density and volume on a CD and not make you want to turn it off after three songs. I don't know how to put it in print in a diplomatic way, but when you get mastering engineers together and you get a couple of beers in them, they'll all agree that CDs are too loud. We hate it and wish we didn't have to do it, then it's right back to work on Monday and squeeze the shit out of it all over again.

GLENN MEADOWS: *The level wars? We had level wars in vinyl right near the end of it, where everybody was trying to get the vinyl hotter and hotter and hotter. And at least in vinyl you had this situation where when the record skipped, the record label would say, "Well, it's too loud, and you're gonna have returns." We originally thought we had that type of limitation on digital, but what ended up happening is there's so many tools out now for doing the dynamic range squash that you can literally get tracks now where you put them in a workstation and it looks like a 2 by 4. It comes on at the quietest passage on the beginning of the intro and it's full level. You get into what I call "dynamics inversion." Spots in the record that should get louder actually get softer because they're hitting the compressor/limiter too hard. I don't think that the record companies and the producers at this point have enough insight or understanding about what radio has learned a long time ago, which is the tune-out factor for distortion.*

GREG CALBI: *It's gotten so insane. I'm a huge music fan and I listen to CDs constantly at home. I have to say that the CDs that always please me the most sonically are not the real hot ones when I bring them in here and look at them on the meters. I tell people, "If you want yours to be hot, I know how to do it, and I'll make it as hot as we can possibly make it and still be musical. But I just want to tell you that you may find that it's not as pleasing to you if you get it too hot."*

BERNIE GRUNDMAN *I just don't think that you should do anything that draws attention to itself. Like if you're going to use a compressor or limiter on the bus, if you use it to the point where you really hear a change in the sound, you're going a little too far. Some of the automatic settings in these devices really aren't as good as they make them out to be. And when you use them, you have to realize that you're going to degrade the sound, because compressors and limiters will do that. If you put a compressor in the circuit, not even compressing, you will hear a difference, and it will sound worse.*

But getting the most level onto the disc or file is not the only level adjustment that the mastering engineer must practice. Just as important is the fact that every song on the disc must be perceived to be just as loud as the next. Once again, *perceived* is the key word, since this is something that can't be directly measured and must be done by ear.

How to Get Hot Levels

The bulk of the audio-level work today is done by a combination of two of the mastering engineer's primary tools—the compressor and the limiter, which, contrary to recording practices where there's one box that can do either job (depending on the settings), are actually two different boxes in mastering. The compressor is used to increase the small and medium level signals, whereas the limiter controls the instantaneous peaks. Remember, though, that the sound of the compressor and limiter will have an effect on the final audio quality—maybe for the worse—especially if you push them hard.

LIMITING

To understand how a limiter works in mastering, you have to understand the composition of a typical music program first. In general, the highest peak of the source program determines the maximum level that can be achieved from a digital signal. But because many of these upper peaks are of very short durations, they can usually be reduced in level by several dB with minimal audible side effects. By controlling these peaks, the entire level of the program can be raised several dB, resulting in a higher average signal level.

Most digital limiters used in mastering are set as *brick-wall* limiters. This means that no matter what happens, the signal will not exceed a certain predetermined level, and there will be no digital overs. Thanks to the latest generation of digital limiters, louder levels are easier to achieve than ever before because of more efficient peak control. This is thanks to the "look-ahead" function that almost all digital limiters now employ. Look-ahead delays the signal a small amount (about two milliseconds or so) so that the limiter can anticipate the peaks in such a way that it catches the peak before it gets by. Analog limiters don't work nearly as well because an analog input can't predict its input like a digital limiter with look-ahead can. Because there is no possibility of overshooting, the limiter then becomes known as a brick-wall limiter.

By setting a digital limiter correctly, the mastering engineer can gain at least several dB of apparent level just by the simple fact that the peaks in the program are now controlled.

EDDY SCHREYER: *When a program is mixed with a hot snare, for example, I can use a digital limiter that will sort of clip the peak off that so I can back off the dynamics of that particular instrument in the mix without EQing it out. Because if I go for the snare with EQ, I'm going to be pulling down the vocals and possibly the guitars as well. If I go for a kick that's mixed too hot, adjusting 80, 60, 40 cycles or something to pull a kick down, it will really sacrifice the bottom quite a bit, so I tend to use digital limiting to peak limit excessive dynamics in those particular cases.*

COMPRESSION

As the name implies, compression actually increases the lower-level signals, while limiting decreases the loud ones.

Mastering Compressor Tips and Tricks

Following are some tricks and tips for mastering compressors:

- Gain changes on the compressor caused by the drums can pull down the level of the vocals and bass and cause overall volume changes in the program.
- Slower release settings will usually keep the gain changes less audible, but will also lower the perceived volume.
- A slow attack setting will tend to ignore drums and other fast signals, but will still react to the vocals and bass.
- A slow attack setting might also allow a transient to overload the next piece of equipment in the chain.
- If the source is too percussive or has loud drums in the mix, try adjusting the attack and release controls.
- Sometimes fast attack and medium release help tame drums.
- Fast attack and release settings tend to reduce transients.
- Usually only the fastest settings can make a unit pump.
- Slower release settings tend to be the most inaudible.
- The more bouncy the level meter seems, the more likely that the compression will be audible.
- Generally speaking, the trick with compression in mastering is to use a slow release and less (usually much less) than 5 dB of compression.
- Quiet passages that are too loud and noisy are usually a giveaway that you are seriously over-compressing.

The key to getting the most out of a compressor is the *attack* and *release* (sometimes called *recovery*) controls, which have a tremendous overall effect on a mix and therefore are important to understand. Generally speaking, transient response and percussive sounds are affected by the *attack* control setting. *Release* is the time it takes for the gain to return to normal or zero gain reduction.

In a typical pop-style mix, a fast attack setting will react to the drums and reduce the overall gain. If the release is set very fast, then the gain will return to normal quickly, but can have an audible effect of reducing some of the overall program level and attack of the drums in the mix. As the release is set slower, the gain changes that the drums cause might be heard as *pumping*, which means that the level of the mix will increase and then decrease noticeably. Each time the dominant instrument starts or stops, it pumps the level of the mix up or down. Compressors that work best on full program material generally have very smooth release curves and slow release times to minimize this pumping effect.

GLENN MEADOWS: *My typical approach is to use like a 1.15:1 compression ratio and stick it down at –20 or –25 so you get into the compressor really early and you don't notice it going from linear to compressed and basically just pack it a little bit tighter over that range. I'll get maybe 3 dB of compression, but I've brought the average level up 3 or 4 dB, and it just makes it bigger and fatter. People think that they have to be heavily compressed to sound loud on the radio, and they don't.*

Three Rules for Hot Levels

- Set a digital limiter to contain peaks, as mentioned earlier.
- Set a compressor at 1.5:1 or 2:1 to gain apparent level.
- Set your master fader to –1 dB to avoid digital overs.

EDDY SCHREYER: *You go as loud as you can and you begin listening for digital clipping, analog grittiness, and things that begin to happen as you start to exceed the thresholds of what that mix will allow you to do, in terms of level. Again, just spanking as much gain as you can, be it in the analog or digital world, doesn't matter. You go for the level and properly control it with compression, then you start to EQ to achieve this balance. Of course, it all depends on the type of mix, how it was mixed, the kind of equipment that was used, how many tracks, the number of instruments, and the arrangement.*

GREG CALBI: *What I do in general is try to use three or four different devices to a point where each one is just a little past the point of overload. I overdrive two, sometimes three, and even four pieces of gear, one of them being an A-to-D converter, and the other ones being digital level controls. I find that if I spread the load out amongst a couple of different units and add them together, then I'm able to get it as loud as I can.*

To Normalize or Not to Normalize

Professional mastering engineers do *not* use the normalization function of a DAW to adjust level. Normalization looks for the highest peak of the audio file and adjusts all the levels of the file upward to match that level. Although that seems like a very simple and easy way to adjust levels, it is seldom, if ever, used. The reason is it really doesn't do as good a job at creating average levels in between songs as the human ear, and, in the worst case, it can degrade the audio quality.

As stated before, what normalization does is look for the highest peak of the audio file and adjust all the levels of the file upward to match that level. Even the smallest adjustment inside the DAW can sometimes cause massive DSP recalculations, all to the detriment of the ultimate sound quality.

But the biggest problem of normalizing is that it just looks at the digital numbers involved and not at the content of the music. As a result you end up with some songs (ballads, for example) that are way too loud because of the way they're electronically boosted.

Ultimately, what we're actually looking for is equal *perceived* loudness between songs, not equal electronic loudness. This is something that normalization can't achieve.

BOB KATZ: *I'll give you two reasons [why I don't normalize]. The first one has to do with just good old-fashioned signal deterioration. Every DSP operation costs something in terms of sound quality. It gets grainier, colder, narrower, and harsher. Adding a generation of normalization is just taking the signal down one generation.*

The second reason is that normalization doesn't accomplish anything. The ear responds to average level and not peak levels, and there is no machine that can read peak levels and judge when something is equally loud.

Frequency Balance

One of the most important charges of the mastering engineer is fixing the frequency balance of a project (if it's needed). Of course this is done with an equalizer, but the type used and the way it's driven are generally far different than during recording or mixing. Whereas in recording you might use large amounts of EQ (from 3 to 15 dB) at a certain frequency, in mastering you almost always work in very small increments (usually in tenths of a dB to 2 or 3 at the very most). What you will see is a lot of small shots of EQ along the audio frequency band, but again in very small amounts.

For example, these might be something like +1 at 30 Hz, +.5 at 60 Hz, .2 at 120 Hz, –.5 at 800 Hz, –.7 at 2500 Hz, +.6 at 8 kHz, and +1 at 12 kHz. Notice that there's a little happening at a lot of places.

Frequency Feathering

Another technique that's used frequently is known as *feathering*. This means that rather than applying a large amount of EQ at a single frequency, you add small amounts at the frequencies adjoining the main one. An example of this would be instead of adding +3 dB at 100 Hz, you add +1.5 dB at 100 Hz and +.5 dB at 80 and 120 Hz. This lowers the phase shift brought about when using analog equalizers and results in a smoother sound.

Four Rules for Frequency Balancing

- Know the sound you're going for.
- Use a little EQ at a time—a little goes a long way.
- Feather the frequencies.
- A/B against the original.

BERNIE GRUNDMAN: *One of the things that is really hard is when the recording isn't uniform. What I mean by uniform is that all of the elements don't have a similar character in the frequency spectrum. In others words, if a whole bunch of elements are dull and then just a couple of elements are bright, then it's not uniform. And that's the hardest thing to EQ because sometimes you'll have just one element, like a hi-hat, that's nice and bright and crisp and clean, and everything else is muffled. That is a terrible situation because it's very hard to do anything with the rest of the recording without affecting the hi-hat. You find yourself dipping*

and boosting and trying to simulate air and openness and clarity and all the things that high end can give you, and so you have to start modifying the bottom a lot. You do the best you can in that situation, but it's usually a pretty big compromise.

DAVE COLLINS: *I guess when we were talking about the philosophy of mastering, what I should have added was that one of the hardest things—and it took me forever to get this—is knowing when to* not *do anything and leave the tape alone. As I have gained more experience, I am more likely to not EQ the tape, or just do tiny, tiny amounts of equalization.*

EDDY SCHREYER: *Frequency balance is making adjustments with compression, EQ, and such so that it maintains the integrity of the mix, yet achieves balance in the high, mid, and low frequencies. I go for a balance that is pleasing in any playback medium that the program may be heard in. And obviously I try to make the program as loud as I can. That still always applies.*

But there are also limiting factors on what balance can be achieved. Some mixes just cannot be forced at the mastering stage because of certain ingredients in a mix. If something is a little bottom-light, you may not be able to get the bottom to where you would really like it. You have to leave it alone so it remains thinner because it distorts too easily.

Processing on Load-In

Depending on the program, the elite mastering engineers may do some of their level adjustments and equalizing outside the workstation and then record that result into the DAW. This is mostly a sonic issue, since the dedicated outboard devices may sound better than what can be offered within the DAW for that particular type of music or program.

Editing

Editing during mastering has gone through a complete metamorphosis in just a few short years. Until the mid-1980s, when most mastering entered the digital age, most editing was still done by hand using a razorblade and splicing magnetic tape on an analog two-track recorder. But as the demand for CDs began to rise, razorblade editing quickly gave way to electronic editing in the digital domain using a Sony DAE-3000, which was basically a modified video editor, and two BVU-800 (and later DMR 4000) 3/4" video decks that carried the digital audio. Today, virtually all editing is done on a digital audio workstation, which is a hardware/software package using a personal computer as the engine.

Although the speed and capability vary from unit to unit, the main operations required by the mastering engineer remain the same. The mastering engineer must supply fades (both fade-ins and fade-outs), basic additions/subtractions to the song via cut and paste techniques, and sometimes spreads (the time between songs). As with most mastering operations, what may seem easy can be enormously difficult without the proper knowledge of how to apply the proper tools.

FADES

Almost anyone with a workstation knows how to apply fades, but does that mean that they are the right fades? Another one of the main elements of professional mastering is making sure that the fade not only happens, but sounds smooth as well. As a result, the mastering engineer is frequently called upon either to do the fade entirely or to help it out. Even in these days of automated mixing and drawn-in fades in the workstation, many mix engineers still actually leave the master fade-out completely up to the mastering engineer.

FADE-INS

There are two schools of thought on the fade-ins or headfades—one uses a sharp "butt cut," and the other uses a more gradual algorithmic fade. Regardless of which type of fade is chosen, the principle is to get rid of count-offs, coughs, and noises left on the recording before the song begins. Although this seems to be an easy procedure, you must use care in order to maintain the naturalness of the downbeat.

BOB KATZ: *At the head of things, it's not as easy. The biggest problem with the headfades is that people just cut them off. The breath at the beginning of a vocal is sometimes very important. But if you cut something—and not just the breath, but something which I guess we would call the air around the instruments prior to the downbeat—it doesn't sound natural.*

FADE-OUTS

The type of fade selection used can make a big difference, as you'll see. The temptation is to use a linear curve to make a fade, as in Figure 4.2. However, an exponential curve (see Figure 4.3) is sometimes smoother and much more realistic sounding.

BOB KATZ: *If you're good at editing, you can supply artificial decays at the end of songs with a little reverb and a careful crossfade that's indistinguishable from real life.*

Figure 4.2
A linear fade.

Figure 4.3
An exponential curve.

BOB LUDWIG: *Oh yeah, it happens often enough. Speaking of Pro Tools, a lot of people assemble mixes on Pro Tools and they don't listen to it carefully enough when they're compiling their mix, and they actually cut off the tails of their own mixes. You can't believe how often that happens. So a lot of times we'll use a little 480L to just fade out their chopped-off endings and extend naturally.*

EDDY SCHREYER: *I've had some projects where they clipped intros and I've had to grab beats from other places and put them on the top, so I prefer it if you don't cut the program too tight.*

Even when a fade is made during the mix, it sometimes needs some help due to some inconsistencies. "Following the fade" means drawing a curve that approximates the one on the mix, only smoother (refer to Figure 4.2).

SPREADS

The *spread* is the time between each song when mastering for CD or vinyl distribution. Although this might seem to be quite arbitrary in many cases, the savvy mastering engineer usually times the spread to correspond with the tempo of the previous song. In other words, if the tempo of the first song was at 123 beats per minute, the mastering engineer times the very last beat of the first song to stay in tempo with the downbeat of the next. The number of beats in between depends upon the flow of the album.

Please note that this might not be appropriate in all cases because each project is unique. It is a place to start, however. Many times a smooth flow between songs is not desirable, and a longer space is far more appropriate. The spread in that case is replaced with a two-, three-, or four-second area in between songs to keep them disconnected.

EDIT DECISION LISTS (EDLS)

Instead of using cut and paste operations to determine the song sequence and spreads, many professional workstations used for mastering use the edit decision list (EDL). The EDL, which was originally developed for video editing, makes it easy to change the order of songs at any time. The EDL is the list of all the elements that make up the final result and the positions those elements will take in the final sequence. Those elements are usually songs and will also be described in some fashion, usually by the name of the song.

Effects

Although mastering engineers have occasionally been asked to add effects through the years, it has now become far more commonplace than ever before. This is partly due to the proliferation of digital audio workstations, where a poorly chosen fade is used prior to mastering. And then there's still the fact that many artists and producers are sometimes horrified to find that the amount of reverb is suddenly less than they remember during the mix.

Most mastering engineers prefer to add any effects in the digital domain, both from an ease-of-use and from a sonic standpoint, so a reverb plug-in like the Audio Ease Altiverb is chosen.

Sometimes this is done by sending the output of the workstation into the effects device, then recording the result back into the workstation on two different tracks. The resultant effects tracks are then mixed in the proper proportions in the workstation. Because this processing is done in the digital domain, an effects device with digital I/O is essential. Among the devices used are the Lexicon PCM 91 and 300, and the TC Electronic M6000 and 3000, although any high-quality processor that operates in the digital domain will do.

BOB KATZ: *A reverb chamber is used surprisingly a lot in mastering to help unify the sound between things. I might use it on five percent of all my jobs. I discovered the Sony V77, which is already obsolete, in Sony's typical way. After you spend a couple of hours fine-tuning it, it can sound just like an EMT.*

BOB LUDWIG: *I do a fair amount of classical music mastering and very often a little bit of reverb is needed on those projects. Sometimes if there's an edit that for some reason just won't work, you can smear it with a bit of echo at the right point and get past it. Sometimes mixes come in that are just dry as a bone, and a small amount of judicious reverb can really help that out. We definitely need it often enough that we've got a 480L in our place, and it gets used probably once every week.*

CHAPTER 5

Preparation for Mastering

Whether you master your final mixes yourself or take them to a mastering engineer, things will go much faster if you prepare for mastering ahead of time. Nothing is as exasperating to all involved as not knowing which mix is the correct one or forgetting the file name. This part of the chapter offers some tips to get you "mastering-ready."

Things to Remember before Mastering

- **Make Sure You Bring the Highest Resolution Mixes You Can** Lossy formats, such as MP3, Windows Media, Real Audio, and even audio CDs, won't cut it and will give you an inferior product in the end. Bring the highest resolution mixes you can and make the other formats after mastering.

- **Don't Over-EQ When Mixing** Better to be a bit dull and let your mastering engineer brighten things up. In general, mastering engineers can do a better job for you if your mix is on the dull side rather than too bright or too big.

- **Don't Over-Compress When Mixing** You might as well not even master if you've squashed it too much already. Hypercompression deprives the mastering engineer of one of his major abilities to help your project. Squash it for your friends. Squash it for your clients. But leave some dynamics for your mastering engineer. In general, it's best to compress and control levels on an individual track basis and not on the stereo buss.

- **Getting Levels to Match Is Not Important** Just make your mixes sound great. Matching levels between songs is one of the reasons you master your mixes.

- **Getting Hot Levels Is Not Important** You still have plenty of headroom even if you print your mix with peaks reaching –10 dB or so. Leave it to the mastering engineer to get the hot levels. It's another reason why you go there.

- **Watch Your Fades** If you trim the heads and tails of your track too tightly, you might discover that you've trimmed a reverb trail or an essential attack or breath. Leave a little room and let the mastering engineer perfect it.

- **Document Everything** You'll make it easier on yourself and your mastering person if everything is well documented, and you'll save yourself some money too. The documentation expected includes any flaws, digital errors, distortion, bad edits, fades, shipping instructions, and record company identification numbers. If your songs reside on hard disk as files, make sure that each file is properly ID'd for easy identification (especially if you're not there).

  Especially don't be afraid to put down any glitches, channel imbalances, or distortion. The mastering engineer won't think less of you if something got away (you wouldn't believe the number of times it happens to everybody), and it's much easier than wasting a billable hour trying to track down an equipment problem when the problem is actually on the mix master itself.

- **Alternate Mixes Can Be Your Friend** A vocal up, vocal down, or instrument-only mix can be a lifesaver when mastering. Things that aren't apparent while mixing sometimes jump right out during mastering, and having an alternative mix around can sometimes provide a quick fix and keep you from having to remix. Make sure you document them properly, though.

- **Check Your Phase When Mixing** It can be a real shock when you get to the mastering studio, the engineer begins to check for mono compatibility, and the lead singer or guitar disappears because something in the track is out of phase. Even though this was more of a problem in the days of vinyl and AM radio, it's still an important point because many so-called stereo sources (such as television) are either pseudo-stereo or only stereo some of the time. Check it and fix it before you get there.

- **Go to the Session If At All Possible** Most engineers and producers will go to the first few sessions when checking out a new mastering engineer to see whether he has the same musical and technical sensibilities. After that, a bond of trust develops, and they will simply

send the mix master with any instructions. That being said, you should go to all of the mastering sessions if possible because things will always sound a bit different (and probably better) than what it sounded like during mixdown. Attending the session also allows for some final creative decisions that only you can make. (For example, "The kick is a little loud; see whether you can deemphasize it a bit." Or, "Let's squash the whole mix a little more to make this tune punchier.")

- **Come Prepared** Make sure all documentation, shipping instructions, and sequencing are complete before you get there. Sequencing (the order in which the tunes appear on the CD or vinyl record) is especially important, and doing this beforehand will save you a bunch of money in mastering time. Many engineers/producers have the mistaken impression that once the final mix is finished, it's off to the mastering studio. There should be one additional session, however, known as the *sequencing session*. This means that you take a day and do any editing that is required (it's cheaper to do it here than during mastering) and listen to the various sequence possibilities. This is really important if you will be releasing in multiple formats, such as CD and vinyl (yes, there are still some diehards), or in different countries or territories because they will probably require a different song order due to the two sides of the record.

- **Have Your Songs Timed Out** This is important for a couple of reasons. First, you want to make sure that your project can easily fit on a CD, if that's your release format. Most CDs have a total time of just under 80 minutes (78:33, to be exact), although it is possible to get an extended-time CD. (But be careful—you may have replication problems.) Obviously the available time decreases if you choose to include additional files on the ROM section of the disc.

- **Vinyl records may be around for a while (in limited quantities), so the following applies if you intend to cut vinyl!** Cumulative time is important because the mastering engineer must know the total time per side before he starts cutting, due to the physical limitations of the disc. You are limited to a maximum of about 25 minutes per side if you want the record to be nice and loud.

  Because you can only have 25 minutes or less on a side, it's important to know the sequence before you get there. Cutting vinyl is a one-shot deal with no undos like on a workstation. It'll cost you money every time you change your mind.

Interview with Gannon Kashiwa

This part of the chapter consists of an interview with Gannon Kashiwa, Digidesign's Professional Products Market Manager. There has always been a difference of opinion on the sound of DAWs, and since there are more Pro Tools systems than anything else, I thought it would be a good idea to get some insight into DAW fidelity directly from the source. So I talked to Gannon Kashiwa of Digidesign to provide a manufacturer's perspective.

What are the common things that you see that cause a decrease in fidelity in the DAW?
GK: There's a bunch of things that people do to degrade their sound. One of the things is over-compressing and using way too much processing in order to get that CD sound too early in the process. I see mixes that are totally squashed and maximized up to the top of the digital word leaving the studio, heading to mastering. There's nothing wrong with putting a mastering chain on your master fader so you can check it out, but if you leave it on, you're not giving the mastering engineer any choices to work with dynamically and sonically. If you pack the word up into the final two bits of a 24-bit word (that's anything hotter than –12 dB FS), there really isn't much left for those guys to do. You can't uncompress something once it's already maximized.

It seems to me that's a holdover from the days of 16-bit, when you needed to get as close to 0 dB FS to keep the signal from getting noisy. That doesn't seem needed today.
GK: Exactly. You've got 24 bits of audio dynamic range to use. That's 144 dB of dynamic range that is available to you. There's no reason to record up in that top two bits (12 dB) and keep the mix there the whole time.

As a matter of fact, if you record everything really hot, then you're going to have to start pulling the channel faders down and the master fader down in order to avoid clipping. I always recommend for people to leave 3 to 6 dB of headroom or even more (depending upon the kind of music) in their recorded files in their mix. Again, if you maximize it out, you don't have the dynamic range later in the game.

Also if you're always working toward 0 dB FS, with highly dynamic material with a lot of fast transients, there's a chance that you're going to have inter-sample clipping that you wouldn't ordinarily see when the waveform gets reconstructed. If you have a couple of samples that are right at 0 dB FS, in between those samples you might have something that's an overage.

I hadn't heard of that. It's really in between the samples?
GK: It's what happens with the reconstruction filter. It's only getting its information at the sample points, but it's possible to clip the reconstructed waveform in between those samples.

Coming back to over-processing, do you have a recommended method for keeping everything as clean as possible?
GK: As I said, part of the sound today is to make things compressed and loud, but I think what people do is over-compress. They're listening to a mastered final mix of a CD, which is already mastered, and comparing what they're doing as they go along in the recording process. People try to get to the finished sound too quickly.

What I recommend is to mix in groups (drums in one group, vocals in another, and so on) and try to distribute any EQ and compression across a number of stages so you're not trying to get any one equalizer or any one compressor to do a huge amount of work. If you distribute it across a couple of different compressors or EQs where nothing is used to its extreme, you'll get a much cleaner result.

So use buss compression and compression on the instruments, but don't work any one of them too hard unless you want that real "effect" kind of sound because the nonlinearities of a compressor are going to become more extreme and more audible as you push it harder. A little bit at a time is the key. Don't work any of the processors or EQs too hard.

One other thing about making a cleaner mix: Filtering makes a difference. Being bright is sometimes not your friend because you have all this stuff that's competing for the air in your mix, so using the low-pass filters and removing some of the high-frequency content sometimes cleans things up considerably. Sometimes you have all this high-frequency garbage that you don't need, and you have to make space for the stuff that really belongs up there.

How about the theory that you degrade the sound if you move the faders off of unity gain?
GK: Ah, total BS. Pro Tools calculates all volume and pan coefficients as 24-bit coefficients. It doesn't matter where your fader is. Whether your fader is down 5 dB or up 5 dB, there's no mathematical or sonic consequence.

Does gain-staging make a difference? Is it like analog, where you can't have the channel or group faders way above the master?
GK: Sure. Extremes in any of those cases will affect the output. You still have to observe good gain structure throughout the system, especially when you're doing heavy processing. Good analog engineering practices are still good digital engineering practices.

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CHAPTER 6

Mastering for CD

Mastering for CD requires the mastering engineer to know far more than the basics of EQ, dynamics, and editing. In fact, a proper and efficient job entails awareness of many items, from dealing with inserting start/stop and identification codes, to understanding the different choices for the master delivery medium, to checking that master medium for errors.

Dither

Dither is a low-level noise signal that is added to the program in order to trim a large digital word into a smaller one. Since the Red Book specification (see Chapter 12, "Internet Delivery Formats") specifies that the word length for an audio CD must be 16 bits, a program with a longer word length (such as the usual 24 bits used in a DAW) must eventually be decreased. Just lopping off the last 8 bits (called *truncation*) degrades the audio, so the dither signal is used to gently accomplish this task. Since word lengths usually expand when a signal undergoes digital signal processing (up to as many as 64 bits), eventually they must be reduced to 16 bits to fit on a CD. An undithered master will have decay trails stop abruptly or a buzzing type of distortion at the end of a fadeout.

All dither is not created equally. There are currently many different algorithms to accomplish this task, with each DAW manufacturer having either their own version or one supplied by a third party. Generally speaking, dither comes in two major flavors—flat and noise-shaped—with the difference being that flat sounds like white noise and therefore makes what it's applied to a tiny bit noisier, while noise-shaped moves much of this injected noise to an audio band beyond where we can hear. Although it seems like using noise-shaped would be a no-brainer, many mastering engineers continue to use flat dither because they claim that it tends to

"pull together" mixes. Plus, if it's a loud track, you'll be hard-pressed to hear it anyway.

One of the most popular is the POW-r dithering technique that has been produced by the POW-r consortium. This consortium is composed of a number of digital audio powerhouses, including Weiss, SADiE, Millennia Media, Z-Systems, and Lake DSP. POW-r is short for *Psychoacoustically Optimized Wordlength Reduction* and was created in an effort to produce the most sonically transparent dithering algorithm possible. POW-r dither comes in three types that are intended for different kinds of music.

The bottom line on dither is that each type can have a different effect on not only the music, but song to song in the same genre of music. It's worth it to try whatever selection is available before settling on a choice.

For a more in-depth look at dither, check out the dither discussion on Bob Katz's Digital Domain website at http://www.digido.com/bob-katz/dither.html.

Rules for Using Dither

- Dither the signal once and only once. Because dither is a noise signal, it will have a cumulative effect if applied more than once. Plus, dither introduced too early in the signal chain can have a very detrimental effect on any subsequent DSP operations that occur afterward.
- Dither only at the end of the signal chain. The time to dither is just before cutting a Red Book CD or making a DAT.
- Try different types. All dither sounds different, and one may be better for a certain type of music than others. The differences are usually subtle, however.

ISRC

ISRC stands for *International Standard Recording Code* and was developed by ISO (*International Organization for Standardization*) to identify sound and audio-visual recordings. It is officially known as International Standard ISO 3901. ISRC is a unique identifier of each recording that makes up the album. If a recording is changed in any way it requires a new ISRC, but otherwise it will always retain the same ISRC independent of the company or format it is in. An ISRC code also may not be reused.

The ISRC is contained in the subcode (Q-channel) of a CD (see the following "PQ Subcodes" section) and is unique to each track. Each ISRC is composed of 12 characters, as shown in Table 6.1.

Table 6.1 ISRC Composition

Length (Chars)	Description
2	Country
3	First owner (allocated by the RIAA)
2	Year of recording (actually the last two digits)
5	Designation code (assigned by the first owner)

Certain circumstances can cause confusion about how to apply the ISRC. Some of these unique situations are clarified by the following:

- Re-mixes—multiple recordings produced even in the same recording session and even without any change in orchestration, arrangement, or artist require a new ISRC per recording.
- Playing time changes requires a new ISRC.
- Processing of historical recordings requires a new ISRC.
- Back catalog requires a new ISRC for the first re-release.
- Recordings sold or distributed under license by another label use the same ISRCs.
- Compilation without editing of individual tracks may use the same ISRCs.

PQ Subcodes

PQ subcodes control the track location and running-time aspects of CD tracks and enable the CD player to know how many tracks are present, where they are, how long they are, and when to change from one track to another. The ability to place PQs is a fundamental and critical part of a mastering session and DAW software. Should a CD player move to the next track? Should it move in half a second or in two seconds? This is what the PQs control. Placing them correctly is one of the main jobs in creating a Red Book standard master (see Chapter 12), which is what every replication plant requires.

When the CD was first developed, there were a lot of CD subcodes and a lot of uses intended for them in taking control of the disc. In addition to the main data channel of a CD (which contains audio or other data), there are eight subcode channels, labeled P to W, interleaved with the main channel on the disc and available for use by CD audio and CD-ROM players.

CD Subcodes

- P Channel indicates the start and end of each track and was intended for simple audio players that did not have full Q-channel decoding.
- Q Channel contains the timecodes (minutes, seconds, and frames), the table of contents or TOC (in the lead in), the track type, and the catalog number.
- Channels R to W are for subcode graphics known as CD+G and CD Text that can accompany the main audio data.

Except in very special circumstances where the rare CD+G (graphics) disc is being made, all subcodes except the P and Q are ignored. However, the PQ codes must be supplied with every master sent to the replicator, and therefore they must be added and/or edited. Among the items that might require editing are general offsets of track ID numbers to help with universal playability (some old players take a few frames to unmute the outputs when starting to play a new track, so you need to have the ID mark happen several frames ahead of first frame of audio), changing song times, and ISRCs. One of the reasons why the Sonic Studio and SADiE DAWs are so popular for mastering is because they have a built-in PQ editor.

Usually a PQ log is printed out and sent with the master to the replicator as a check to ensure that the correct songs and ISRCs have been provided (see Figure 6.1). Also, when making any kind of master, the PQ info is put on the master somewhere separate from the audio so the plant can read it, check it against the PQ log you provide, and use it to cut the glass master (see Figure 6.2).

Figure 6.1
A PQ log.

Client : Test Records

Project: Sample

Title : Sample

Date : March 6, 2007

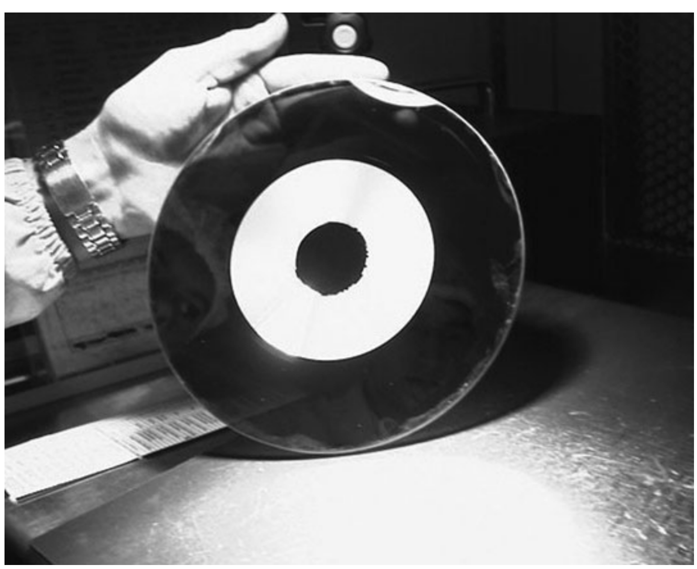
Studio : Test Mastering

Disc Type: Audio
Time Format: 30/NDF
PQ Track 1 Offset: 00:00:00:10 PQ StartOffset: 00:00:00:10
PQ SpliceOffset: 00:00:00:06 PQ EndOffset: 00:00:00:02
PQ MinIndex0Width: 00:00:01:00 UPC/EAN CODE: 0000000000000

PQ Track/Index Information:

T-X	TITLE/ISRC	COPY	EMPH	D/A	NO OFFSET TIME hh:mm:ss:ff	OFFSET TIME hh:mm:ss:ff	OFFSET DURATION hh:mm:ss:ff	CD TIME mm:ss:ff
1	GBCNP7880010	OFF	OFF	A				
0	Pause				00:01:58:00	00:01:57:20	00:00:02:00	00:00:00
1	Suffering & Smiling- Part 1&2				00:02:00:00	00:01:59:20	00:21:31:02	00:02:00
					TOTAL:	00:21:33:02		
2	GBCNP7780130	OFF	OFF	A				
0	Pause				00:23:30:20	00:23:30:22	00:00:02:08	21:33:05
1	No Agreement- Album				00:23:33:10	00:23:33:00	00:15:30:00	21:35:25
					TOTAL:	00:15:32:08		
3	GBCNP7780140	OFF	OFF	A				
0	Pause				00:39:02:28	00:39:03:00	00:00:01:18	37:05:25
1	Dog Eat Dog- Album				00:39:04:28	00:39:04:18	00:15:32:18	37:06:70
					TOTAL:	00:15:34:06		
	LeadOut				00:54:37:04	00:54:37:06		52:39:40
Total					00:52:39:16			

Figure 6.2
A glass master with a CD image in the center. (Image courtesy of DVDBits.com.)



Replication Master Formats

Although pressing plants will routinely accept a common recordable CD as a master, this wasn't always the case, nor is it still the best way. For background's sake, here are a couple of obsolete formats along with the current industry standard.

THE OBSOLETE FORMATS

These next two sections cover the obsolete formats.

The Sony PCM-1630

Time for a bit of history. A longtime staple of the mastering scene (but now very obsolete) is the Sony 1630, which is a digital processor connected to a Sony DMR-4000 or a BVU-800 3/4" U-matic video machine. Once the standard format for the mastering facility to deliver to the replicator, the 1630's 3/4" U-matic tape is noted for its low error count.

The PCM-1630 (its predecessor was the 1610) is a modulation format recorded to 3/4" videotape. It was, for many years, the only way one could deliver a digital program and the ancillary PQ information to the factory for pressing. At the replicator, glass mastering from U-matic can only be done at single speed, so it's usual to transfer the audio data to another media (such as DDP Exabyte) for higher-speed cutting (which is not necessarily a good thing to do from an audio standpoint since the higher the speed of cutting, the greater the error rate).

When directly mastering from U-matic tape, the audio must be recorded at 44.1 kHz to the Sony 1610/1630 format, and the PQ code must be recorded on Channel-1 so that the title can be mastered directly from the U-matic tape. This "PQ burst" (which sounds similar to a modem tone) is basically just a data file placed on the tape before starting the audio.

1630 Mastering Setup

- Digital audio is recorded at 44.1 kHz on the video track.
- Audio begins at two minutes (to eliminate tape-induced errors).
- A PQ code burst is recorded on Audio Track 1 before the digital audio begins.
- Thirty-frame non-drop SMPTE time code is recorded on Audio Track 2.

The PMCD

Another relic from the beginnings of the CD age is the PMCD. PMCD stands for *Pre-Mastered CD* and is a proprietary format jointly owned by Sonic Solutions and Sony. It originally was an effort to replace the Sony PCM-1630 as the standard media delivered to the replicator. It differs from a normal CD-R in that a PQ log is written into the leadout of the disc (see the “How CDs (and DVDs Work)” section later in this chapter). At read-back, this log is used to generate the PQ data set during glass mastering, which eliminates a step during replication. A PMCD can only be written from a Sonic system.

Although a great idea at the time, PMCD never quite lived up to its intentions due to the fact that most masters are now digitally copied to DDP format (see the following section) at the replicator for high-speed glass master cutting. Even though this high-speed cut is faster and more efficient for the replicator, most mastering engineers agree that this high-speed cut sometimes results in an inferior end product thanks to the jitter induced in the process. Current CD-R mechanisms are not capable of creating this format, and plants are no longer equipped to handle PMCD-formatted discs, so the PMCD format has been replaced by DDP.

THE CURRENT STANDARD FORMATS

The following sections discuss some current industry-standard formats.

DDP

There are two ways to deliver your master to a replication facility—audio CD or DDP file. While audio CDs work for this purpose, they are far from ideal because no matter how good the media and the burner are, you’re going to have a number of errors in the data. Disc Description Protocol (DDP) files, however, are delivered as data on a CD-ROM, DVD-ROM, Exabyte tape, or FTP file transmission, and they are the industry-standard method for audio delivery files for replication. The error correction employed by DDP is designed to be more robust than that of audio CDs and ensures that the audio master received by the replicator will have as few errors as possible in the data.

DDP has quickly become a master medium of choice, and there are many reasons why:

- DDP has far fewer errors of any master medium, thanks to computer data quality error correction. CD-Rs and PMCDs have a lot less robust error correction and will output data regardless of whether it’s bad. It’s therefore possible to get different data each time a CD-R is played, and it requires a diligent plant to get an error-free transfer from a CD-R. CD-DA discs, or audio CDs, do not protect the audio data from errors since they assume that the CD player will hide or conceal any errors

during playback. This situation leads to errors in replication when recordable CDs, formatted as Red Book audio discs, are used as replication masters.

- It's easier and safer to go past the 74-minute boundary with DDP. Long CD-Rs are less reliable, although that does not mean they won't work.

- Many plants will transfer a CD-R to DDP before sending it to the glass cutter so that they will be more efficient and cut the glass master at high speed (either 2x or 4x). Although this is better for the plant, it may not sound as good as a single-speed cut.

- It's impossible to play back a DDP without the right equipment, which isn't readily available. This means that there's less chance for an accidental playback of the master, which may damage the medium. A CD-R can get smudged and scratched, but the DDP will stay in its baggie until it hits the plant.

CD-R

Now that DVD/CD recorders are inexpensive and widespread, it's possible to cut a master disc to send to the replicator even without the help of an expensive workstation or piece of software. Most plants now accept common CD-Rs for pressing, but the danger here is that some users may think that they can prepare their own masters without the slightest understanding of what the technical specifications are. Therefore, it's important that we discuss some areas of concern.

Disc-at-Once Mode

To create a disc suitable for pressing, it's important to use what's known as *disc-at-once mode*. This means that the CD-R is cut in one complete pass with no stops where the laser is turned off. Using the other cutting mode, *track-at-once*, is not permitted because it stops the laser between songs, which creates unreadable frames that will cause the disc to be rejected at the plant.

Recorder Speed

High-speed (from 12x to 52x, meaning the recorder is cutting at 12 to 52 times real time) CD-R recording is generally far less desirable than 2x to 8x recording. This is because high-speed recording generally results in greater disc errors and increased jitter. Also, recording power does not increase linearly with speed; therefore, higher-speed recording can reduce the total energy required for recording.

It's also generally acknowledged that discs cut at 1x usually sound better as well. Although there are many theories as to why this occurs, it's widely accepted that the jitter begins to rise with the speed of the cut.

CD-R Recording Tips

- Always use disc-at-once mode.
- Also use the lowest burner speed for the best sound and the lowest number of errors.

Error Checking

Errors on any media are extremely critical because they could make the difference between making a good cut on the glass master and making a reject. These errors can be in many forms, from tape dropouts, to scratches or dust on the tape or disc, to just plain bad media. Therefore, most major mastering facilities use several ways to check whether errors have occurred. Way back when, the Sony DTA-2000 Error Checker was normally used for 1630 projects. For CD-R and PMCD, a measurement unit like the StageTech EC2 is used. If a disk image is being sent via FTP, the mastering DAW usually has a utility to verify the file.

Error rate measurements for discs such as the E series (E11, E21, E31, E12, E22, E32) and BLER provide vital information as to the general condition of a tape or disc. BLER, which is an abbreviation of *BLock Error Rate*, represents frame error rate and is one of the most widely used error measurements. One frame represents the smallest integral data package on a disc and contains 24 bytes of data along with sync, subcode, Q parity, and P parity. Data is read from a CD-ROM at the rate of 7,350 frames per second in a 1x player. BLER measures the rate of bad frames that contain one or more read errors. If 1 percent of the frames contain errors, then BLER will be 73.5 per second at 1x. It is required that a disc have a frame error rate less than 3 percent, or a 1x BLER of 220 per second. High-quality discs have much lower frame error rates than this, usually in the range of 20 to 30.

FTP Transmission

Almost all replicators will now accept master files via FTP (*File Transfer Protocol*). In fact, many prefer to receive your master that way. When using FTP, the best thing to send is a DDP file, since it already contains the necessary error correction to protect against transmission errors.

All replicators will have a secure portion of their website dedicated for FTP transfers. After you place your order, they'll send you the host name, user ID, and password.

Parts Production

Although the more high-profile and documented part of mastering lies in the studio with the mastering engineer, the real bread and butter of the business happens after the fact, during what's known as *production*. Production is the time when the various masters are made, verified, and sent to the replicator. Although it's not a very glamorous portion of the business, it's one of the most important nonetheless, since a problem here can negate a perfect job done beforehand.

Once upon a time, production was a lot more extensive than it is today. For instance, in the days of vinyl, a lot of masters had to be made because a pair (one for each side of the disc) had to be sent to a pressing plant in each area of the country, and overseas if it was a major release. When you consider that every master had to be cut separately with exactly the same process, you can see that the bulk of the mastering work was not in the original rundown, but in the actual making of the masters (which was very lucrative for the mastering house). Over the years the parts production has dwindled to the point that we're at today, where digital copies are a snap to make.

Multiple Masters

Generally speaking every project will have a number of masters cut, depending upon the marketing plans and policy of the label. This usually breaks down as follows:

- **The CD master.** This is the master from which the glass master at the plant will be cut; the glass master will, in turn, ultimately make the replicated CDs. If an artist is to have a worldwide release, a separate master is made for each region of the world.

- **The cassette master.** If the label is going to make cassettes (surprisingly, some still do), then a separate master is required because the song sequence is usually different from the CD due to the split sides of the cassette format. This master is sometimes just a CD with 30 seconds of dead space to indicate a side switch.

- **The vinyl master.** If a record is desired, once again a separate master is required due to the song sequence of the two-sided format.

- **The online master.** Because the online portion of sales is now such a large part of the overall sales picture, a separate MP3 and/or AAC master is made. This master is specially tweaked to provide the best fidelity with the least amount of bandwidth.

- **Backup masters.** Most major labels will ask for a backup master that they will store in the company vault. Many times the mastering facility will make a "house" backup as well to save time should a new master be required at a later date.

Client Refs

Before production occurs, a reference, or *ref*, is made for the client to approve. In the vinyl days this was actually a record known as an *acetate*, but now it is more likely to be an MP3 or a CD-R. The client can then take the MP3 or CD to an environment that he's comfortable with and approve the edits, fades, EQ, compression, sequencing, and general sound quality. Any changes will be relayed back to the mastering engineer, who will make those changes and cut another ref for the client to approve. As soon as the client is satisfied, the process of production begins.

Master Verification

Before a master is sent to the replicator, it is verified several different ways to ensure sonic integrity. If using a CD-R, the disc may be tested using a StageTech EC2. If the BLER rate exceeds 220, the disc must be rejected, although it is usually rejected far before that rate. Acceptable BLER rates usually range from 20 to 30 per second.

Most major pressing facilities will also employ some type of audio verification as well, for which a production engineer will listen to the contents of the master (sometimes with headphones) to ensure that it is free from pops, clicks, or glitches.

This attention to detail, as well as the large number of man hours required to create and verify a master, enables the mastering house to charge a premium for this service. A master sent to the replicator can range anywhere from $350 to $1,000, depending on the mastering facility.

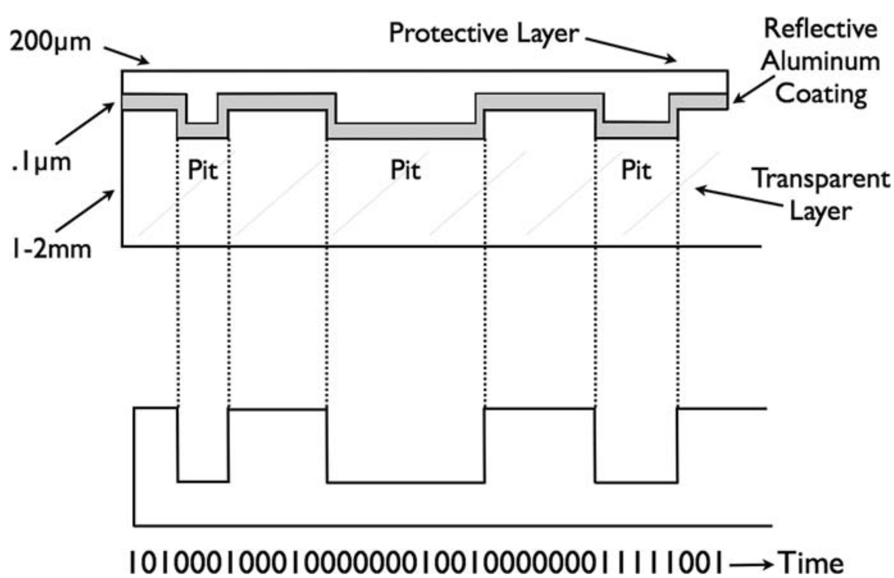
CD Replicators

- Sonopress. 108 Monticello, Weaverville, NC 28787. www.sonopress.com.
- Cinram. 1600 Rich Road, Richmond, IN 47374. www.cinram.com.
- Cinram. 4905 Moores Mill Rd, Huntsville, AL 35811-1511.
- Cinram. 1400 East Lackawanna Avenue, Olyphant, PA 18448.
- Sony DADC. (800) 358-7316. www.sdm.sony.com.
- Amtech. 716 Golf Road, Nuns Island, QC Canada, H3E 1A8. (800) 777-1927.

How CDs (and DVDs) Work

Everything in this section also applies to DVDs, but I'll just use CDs as an example. A CD is a plastic disc 1.2mm thick and 5 inches in diameter that consists of several layers. First, to protect the microscopically small pits (more than 8 trillion of them) against dirt and damage, the CD has a plastic protective layer on which the label is printed. Then there is an aluminum coating that contains the ridges and reflects laser light. Finally, the disc has a transparent carrier through which the actual reading of the disc takes place. This plastic forms a part of the optical system (see Figure 6.3).

Figure 6.3
The CD has several layers. Notice how the ridges contain binary information.



Mechanically, the CD is less vulnerable than a record, but that does not mean that it must not be treated with care. Since the protective layer on the label side is very thin (only one ten-thousandth of an inch), careless treatment or granular dust can cause small scratches or hair cracks, enabling the air to penetrate the evaporated aluminum coating. If this occurs, the coating will begin to oxidize.

The reflecting side of the CD is the side that is read. People tend to set the CD down with the reflecting side up. However, the more vulnerable side is not the reflecting side, but the label side. On the label side, the reflecting layer with its ridges has been evaporated. The sensitive layer on the reflecting side has been protected better than the one on the label side. It is therefore better to store CDs with the reflecting side down. It is best to store the CD back in the jewel case, where it is safely held by its inside edge.

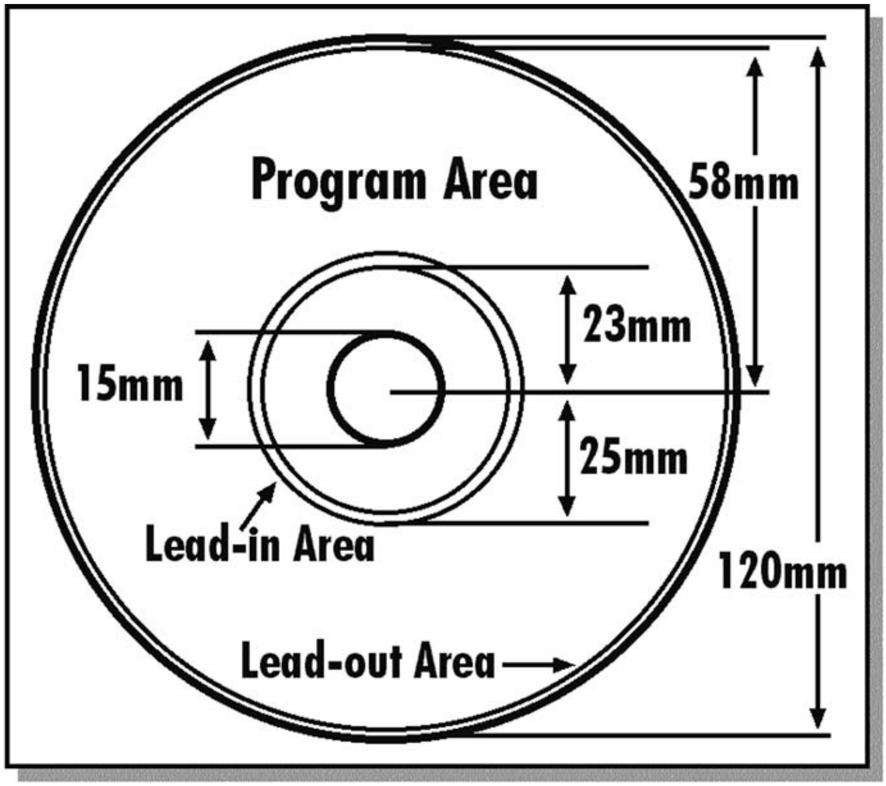
Never write on the label side, even with a felt-tipped pen. The ink may penetrate the thin protective coating and affect the aluminum layer.

CDs are easily scratched and should only be cleaned with a soft cloth, which should be cleaned radially—not along the grooves, but at right angles to the direction of the grooves. If a smear, however small, should remain on the CD, running along the direction of the grooves, much information could be lost. It is advisable to use special CD cleaner that operates with a rotating brush at right angles to the direction of the grooves.

The area of the disc that contains data is divided into three areas (see Figure 6.4):

- The *lead-in* contains the table of contents in the subcode Q-channel and allows the laser pickup head to follow the pits and synchronize to the audio or computer data before the start of the program area. The length of the lead-in is determined by the number of tracks stored in the table of contents.
- The *program* area contains up to about 76 minutes of data divided into a maximum of 99 tracks.
- The *lead-out* contains digital silence or zero data and defines the end of the CD program area.

Figure 6.4
The CD layout.



Scanning the Disc

Like vinyl records, the information on optical discs is recorded on a spiral track in the form of minute indentations called *pits*. These pits are scanned from the reverse side of the disc (this makes them to appear as ridges to the laser) by a microscopically thin red laser beam during playback. The scanning begins at the inside of the back of the disc and proceeds outward. During playback, the number of revolutions of the disc decreases from 500 to 200 rpm (*revolutions per minute*) in order to maintain a constant scanning speed. The disc data is converted into electrical pulses (the bit stream) by reflections of the laser beam from a photoelectric cell. When the beam strikes a land, the beam is reflected onto a photoelectric cell. When it strikes a ridge, the photocell will receive only a weak reflection. A D/A (*digital-to-analog*) converter converts these series of pulses to binary coding, then to decimal values, and then back to an analog waveform (see Figure 6.5).

It should be noted that the ends of the ridges seen by the laser are ones, and all lands and ridges are zeros. Thus, turning on and off the reflection is one, while a steady state is a string of zeroes.

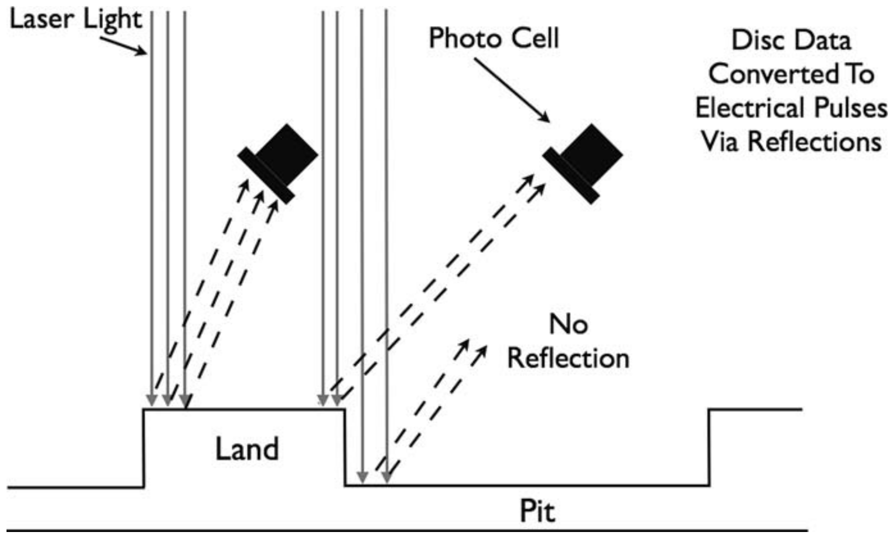


Figure 6.5
The disc data is converted into electrical pulses (the bit stream) by reflections of the laser beam off a photoelectric cell.

Thanks to this optical scanning system, there is no friction between the laser beam and the disc. As a result, the discs don't wear regardless of how often they're played. Discs must be treated carefully, however, since scratches, grease stains, and dust could diffract the light and cause some data to be skipped or distorted. This problem is solved by an error-correction system that automatically inserts any lost or damaged information. Without this error-correction system optical disc players would not have existed, because even the slightest vibration would cause sound and image distortions.

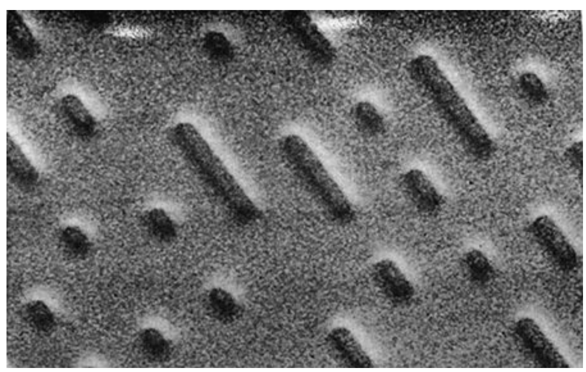


Figure 6.6
A microscope look at CD pits and land. (Image courtesy of Philips, inventors of the CD.)

The scanning must be very accurate because the track of ridges is 30 times narrower than a single human hair. There are 20,000 tracks on one compact disc (refer to Figure 6.3). The lens, which focuses the laser beam on the disc, has a depth of field of about 1 micron (micrometer = one-millionth of a meter).

It is quite normal for the disc to move back and forth 1mm during playback. A flexible regulator keeps the lens at a distance of +/–2 microns from the rotating disc. For the same reason, a perfect tracking system is

required. The complex task of following the track is controlled by an electronic servo system. The servo system ensures the track is followed accurately by measuring the signal output. If the output decreases, the system recognizes this as being "off track" and returns the tracking system to its optimum state.

Many CD players use three-beam scanning for correct tracking. The three beams come from one laser. A polarized prism projects three spots of light on the track. It shines the middle one exactly on the track, and the two other "control" beams generate a signal to correct the laser beam immediately, should it deflect from the middle track.

How CDs Are Made

Data is copied onto the CD in a "pit-and-land" pattern that begins at the inner hub of the disc and spirals toward the outer edge in a counterclockwise direction. For the typical 700 megabytes of data, the continuous track is more than four miles long.

The data is represented by a series of pits and lands that are so small that the width of a human hair would cover more than 40 tracks. More than 60 CD tracks could be placed within a single LP groove.

STEP 1

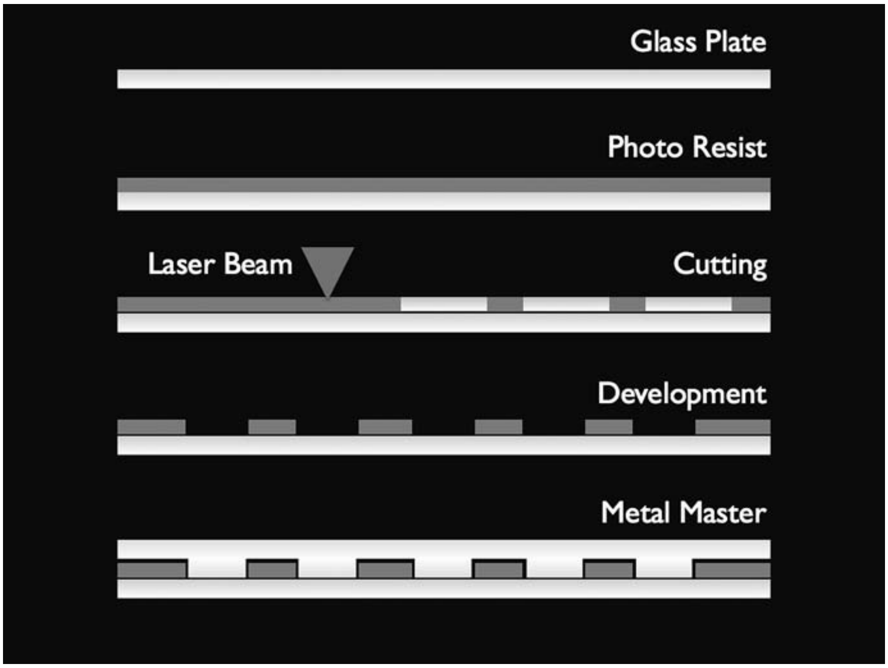
The first step, glass mastering, is composed of a number of stages that are required to create a metallized glass master from which CD stampers are produced. All of the processes are carried out in a clean room, where the mastering technicians must wear special clothing, such as facemasks and footwear, to minimize any stray particles.

The 8" diameter, 6mm thick glass blanks can be recycled, so glass master preparation begins by stripping the old photo-resist from its surface, which is then followed by a washing with de-ionized water and then a careful drying. The surface of the clean glass master is then coated with a photo-resist layer a scant 150 microns thick with the uniformity of the layer measured with an infrared laser. The photo-resist coated glass master is then baked at about 176 degrees for 30 minutes, which hardens the photo-resist layer and makes it ready for exposing by laser light.

Laser-beam recording is where the photo-resist layer is exposed to a blue gas laser fed directly from the source audio of a DDP master tape or file. The photo-resist is exposed where pits are to be pressed in the final disc. The photo-resist surface is then chemically developed to remove the photo-resist exposed by the laser and therefore create pits in the surface.

These pits then extend right through the photo-resist to the glass underneath to achieve the right pit geometry. The glass itself is unaffected by this process (see Figure 6.7).

Figure 6.7
Making the glass master.



STEP 2

The surface of this glass master, which is called the *metal master* or *father*, is then coated with either a silver or a nickel metal layer. The glass master is then played on a disc master player (DMP) to check for any errors. Audio masters are actually listened to at this stage.

STEP 3

The next stage is to make the reverse image stamper, or *mother* (a positive image of the final disc pit and land orientation). The mother is then form pressed onto the extruded "children" membranes, which ultimately contain all the binary information used to play the disc.

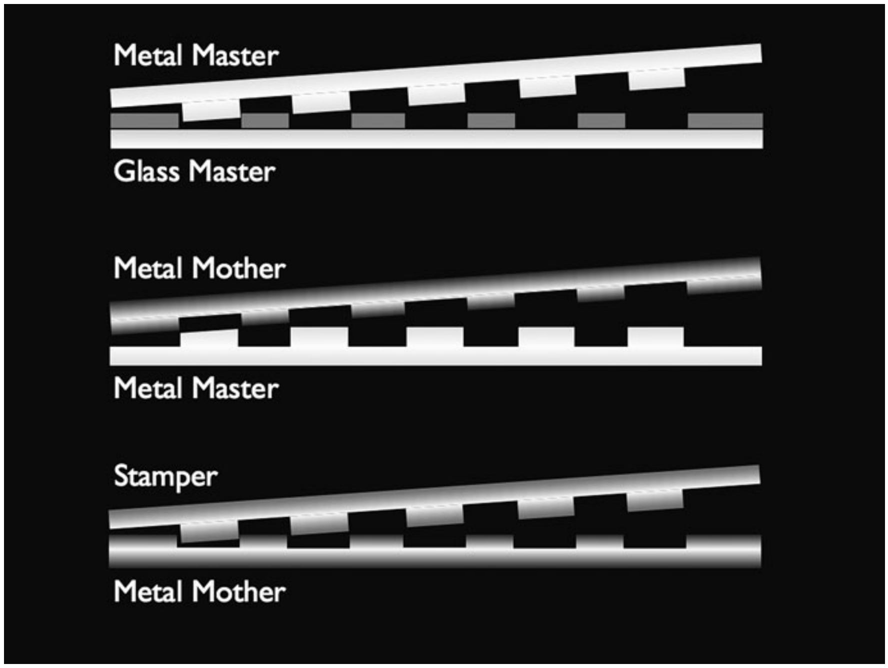
STEP 4

Stampers are then made from the mother (negative image) and secured into the molding machines that actually stamp the CD discs (see Figure 6.8).

STEP 5

After a CD disc has been molded from clear polycarbonate, a thin layer of reflective metal is bonded onto the pit and land surface, and a clear protective coating is applied.

Figure 6.8
Making the stamper. (Diagram courtesy of Sony Disc Manufacturing.)



STEP 6

The disc label is printed on the non-read surface of the disc, and the CD is inserted into a package, such as a jewel case with tray, booklet, and backliner.

A single unit called a *Monoliner* (see Figures 6.9 and 6.10) is actually used to replicate CDs after the stamper has been created. The Monoliner consists of a complete replication line composed of a molding machine, a metalizer, a lacquer unit, a printer (normally three-color), and inspection. Good and bad discs are transferred to different spindles. Finished discs are removed on spindles for packing. It's also possible for the Monoliner to not include a printer so a new job can continue without being stopped while the printer is being set up.

A Duoline is a replication line composed of two molding machines, a metalizer, a lacquer unit, and inspection. Each molding machine can run different titles, with the discs being separated after inspection and placed on different spindles.

Figure 6.9
A Monoliner.

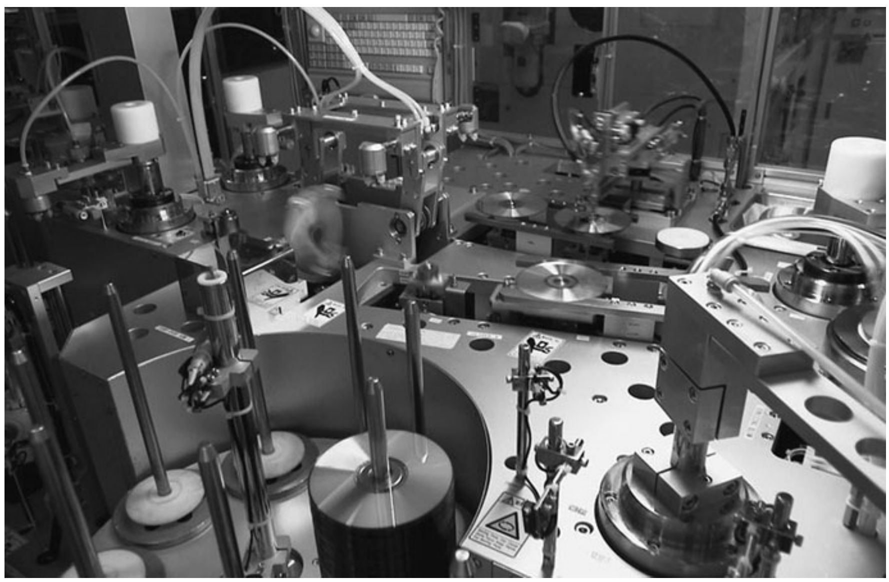


Figure 6.10
A Monoliner in action.

Of Additional Interest

If you need even more information about CDs, go to Chapter 12 or take a look at the following websites.

- Andy McFadden's CD Recordable FAQ: www.cdrfaq.org
- The CD Information Center: www.cd-info.com
- Doug Carson Associates: www.dcainc.com

CHAPTER 7

Mastering for Vinyl

Although it seems like almost an ancient technology in these days of ones and zeroes, the vinyl record seems to be at least holding its own in the marketplace, and even making a bit of a resurgence. This is in no small part due to the high demand from DJs, but also from an audiophile community that still insists that vinyl packs a sonic punch second to none.

Although it's pretty certain that most engineers won't be getting the gear to do vinyl anytime soon, it's still pretty important to know what makes the format tick in order to get the best performance if you decide to make some records along with the CDs. But before we get into the mastering requirements for vinyl, let's take a look at the system itself and the physics required to make a record. Although this is by no means a complete description of the entire process of cutting a record, it is a pretty good overview.

DAVID CHEPPA: *If you just want to cut a mediocre record, you don't need to know a lot of anything. If you want to cut a better record, it's good to know something. If you want to cut an incredible record, you need to have an understanding of the physical world and the physical laws that govern it. You have to know what the limits really are, physically and electronically. So I think it's a balance of art, science, and technology.*

A Brief History of Vinyl

It's important to look at the history of the record because in some ways it is the history of mastering itself. Until 1948, all records were 10" and played at 78 RPM. When Columbia Records introduced the 12" 33 1/3rd RPM in 1948, the age of high-fidelity actually began, since the sonic quality took a quantum leap over the previous generation of disk. However,

records of that time had a severe limitation of only about 10 minutes of playing time per side because the grooves were all relatively wide in order to fit the low frequencies on the record.

To overcome this time limitation, two refinements occurred. First, the Recording Industry Association of America (RIAA) instituted an equalization curve in 1953 that narrowed the grooves, thereby allowing more of them to be cut on the record, which increased the playing time and decreased the noise. This was done by boosting the high frequencies by about 17 dB at 15 kHz and cutting the lows by 17 dB at 50 Hz when the record was cut. The opposite curve is then applied during playback. This is what's known as the *RIAA curve*. It's also the reason why your turntable sounds so bad when you plug it directly into a mic or line input of a console. Without the RIAA curve, the resulting sound is thin and tinny due to the overemphasized high frequencies and attenuated low frequencies.

The second refinement was the implementation of variable pitch, which allowed the mastering engineer to change the number of grooves per inch according to the program material. In cutting parlance, pitch is the rate at which the cutter head and stylus travel across the disk. By varying this velocity, you can vary the number of grooves as well. These two advances increased the playing time to the current 25 minutes or so per side.

In 1957 the stereo record became commercially available and really pushed the industry to the sonic heights that it has reached today.

The Physics of Vinyl

To understand how a record works, you really must understand what happens within a groove. If you were to cut a mono 1-kHz tone, the cutting stylus would swing side to side in the groove 1,000 times per second (see Figures 7.1 through 7.14). The louder you want the signal, the deeper you have to cut the groove.

Although this works great in mono, it doesn't do a thing for you in stereo, and in fact this was a problem for many years. As stated before, stereo records were introduced in 1957, but the fact of the matter is that the stereo record-cutting technique was actually proposed in 1931 by famed audio scientist Alan Blumlein. His technique, called the 45/45 system, was revisited some 25 years later by the Westrex Corporation (who were the big guns in record equipment manufacturing at the time) and resulted in the eventual introduction of the stereo disk.

Essentially, a stereo disk combines the side-to-side (lateral) motion of the stylus with an up-and-down (vertical) motion. The 45/45 system rotated the axis 45 degrees to the plane of the cut. This method actually has several advantages. First, mono and stereo disks and players become totally compatible, and, second, the rumble (low-frequency noise from the turntable) is decreased by 3 dB.

The following figures and accompanying information are courtesy of Clete Baker at Studio B in Lincoln, Nebraska, and detail what the record grooves can look like under different signal conditions.

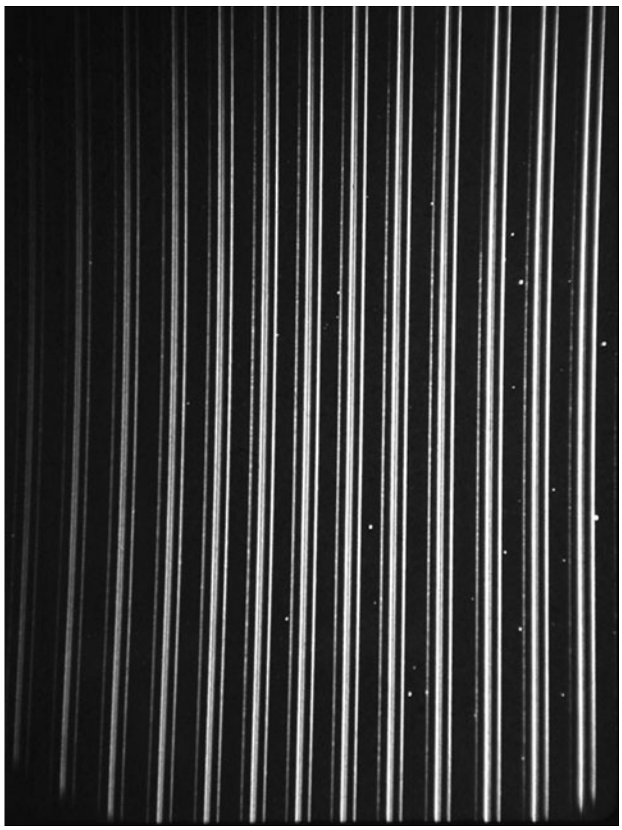


Figure 7.1
A silent groove with no audio information. The groove width across the top of the "vee" from land to land is 2 mils (.002 inch) as measured with the microscope's graticule, which had to be removed for the camera mount. Groove depth is approximately the same as the width for this particular stylus (Capps).

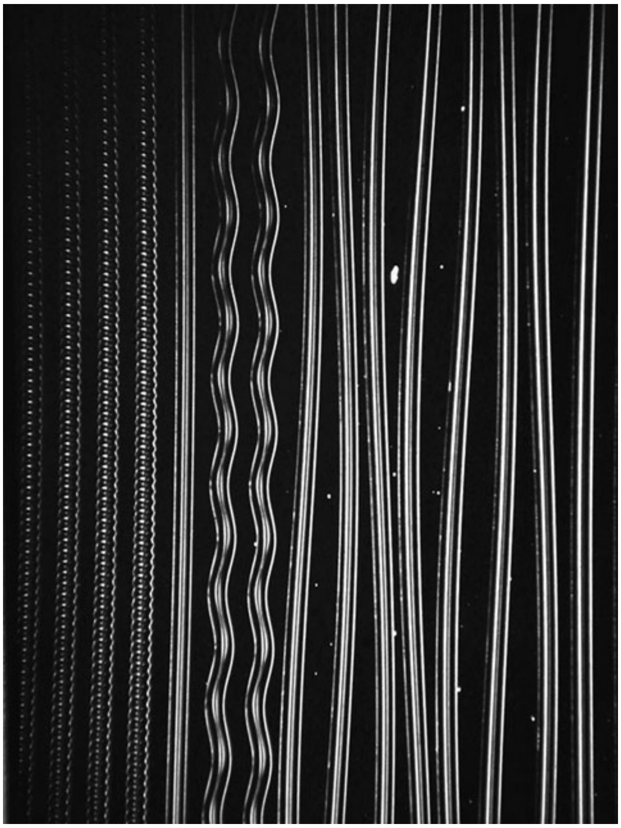


Figure 7.2
From the outside diameter in: A low-frequency sine wave, a mid-frequency sine wave, and a high-frequency sine wave, all in mono (lateral excursion). All frequencies were at the same level at the head end of the system (in other words, prior to application of the RIAA curve). This demonstrates that for any given level, a lower frequency will create a greater excursion than a high frequency, and thus will require greater pitch to avoid intercuts.

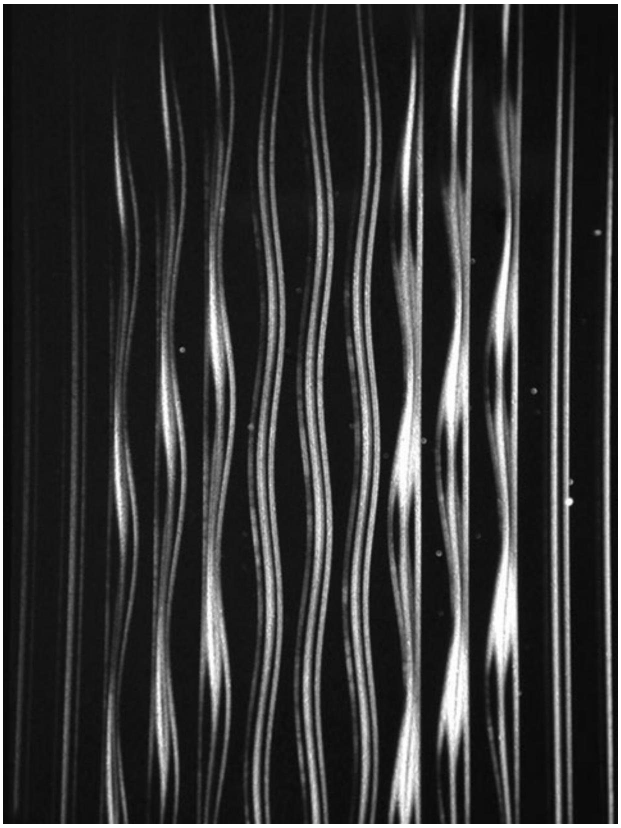


Figure 7.3
This is a sine wave applied to left channel only toward the outer part of the record, summed to mono in the center of the view, and applied to right channel only toward the inner part of the record. One can easily see the difference between the purely lateral modulation of the mono signal and the vertical of the left and right channel signals.

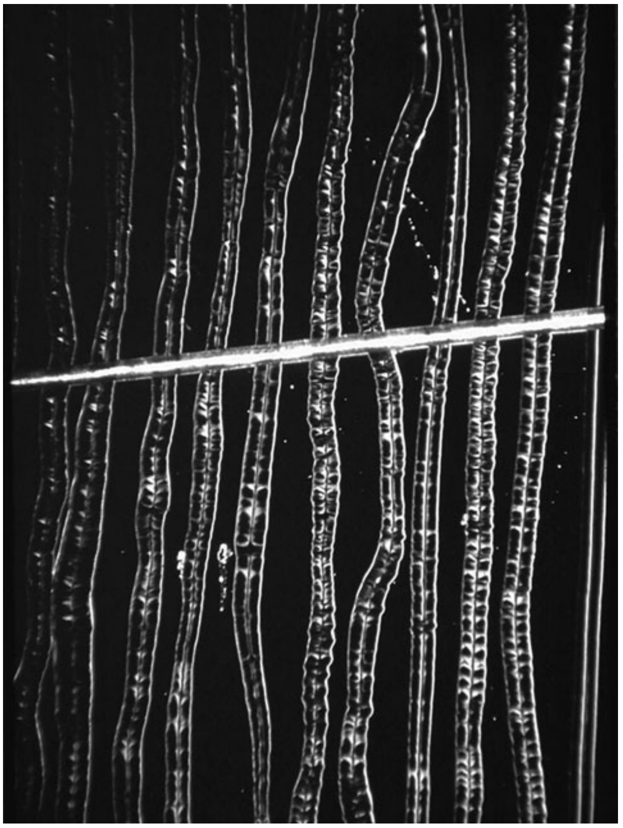


Figure 7.4
A human hair laid across the groove offers a point of reference for size.

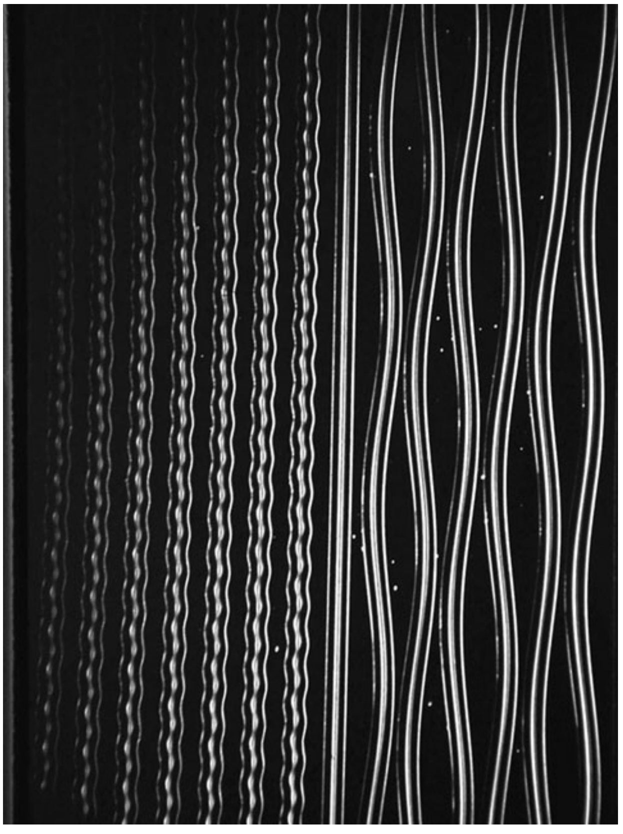


Figure 7.5
Again, lower-frequency and higher-frequency sine waves demonstrate that more land is required to accommodate the excursion of lows than of highs.

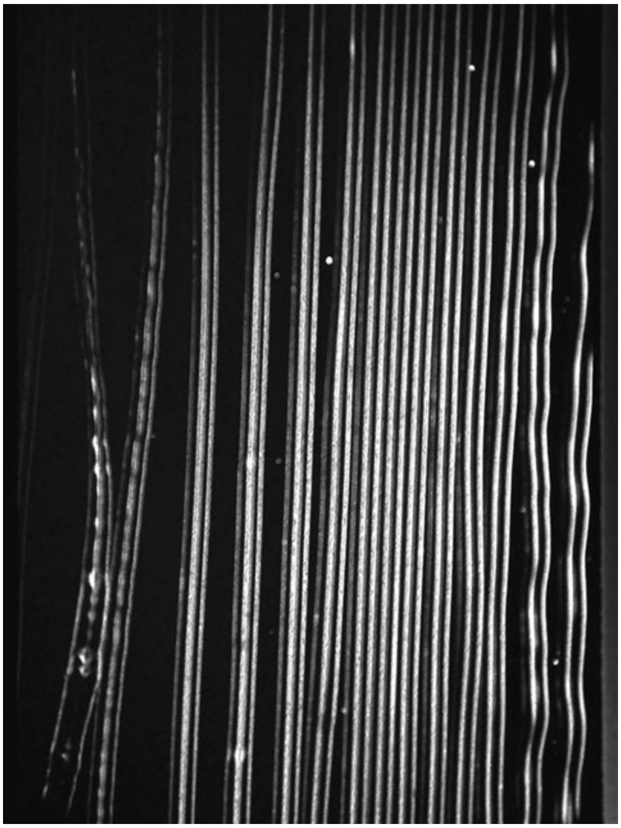


Figure 7.6
To allow for the accommodation of low-frequency excursions without wasting vast amounts of disk real estate, variable pitch is employed to spread the groove in anticipation of large excursions and narrow the groove in the absence of material that doesn't require it. This figure shows variable pitch in action on program audio.

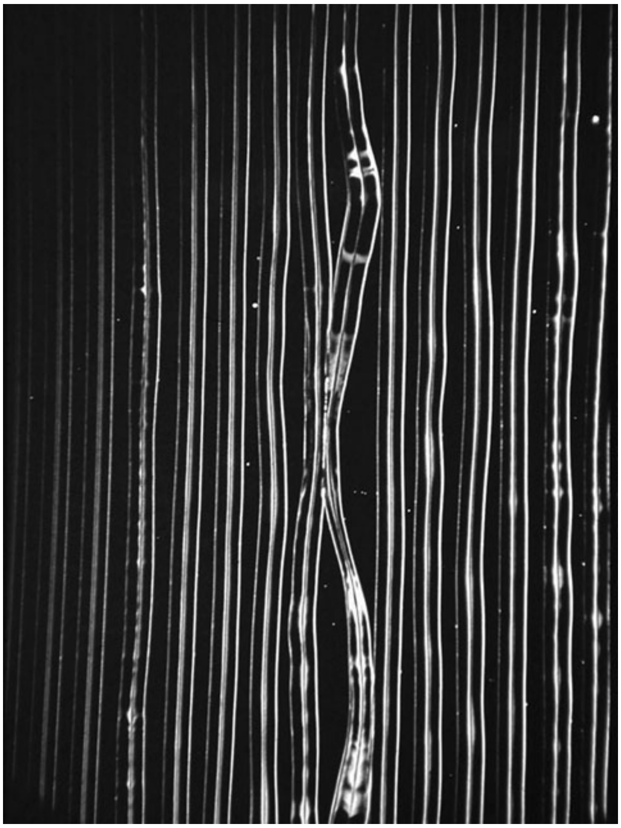


Figure 7.7
When variable pitch goes bad. Oops...a lateral intercut caused by insufficient application of variable pitch for a wide lateral excursion. Toward the bottom center of the slide the outside wall of the loud low frequency has "kissed" the adjacent wall of the previous revolution, but the wall has not broken down; at least 2 mils of depth separates the two, which is a safe margin. However, on the next revolution a chance excursion toward the outside of the disk has all but overwritten its earlier neighbor; less than half a mil separates the bottom of the grooves there, which is certain to cause mistracking down the line.

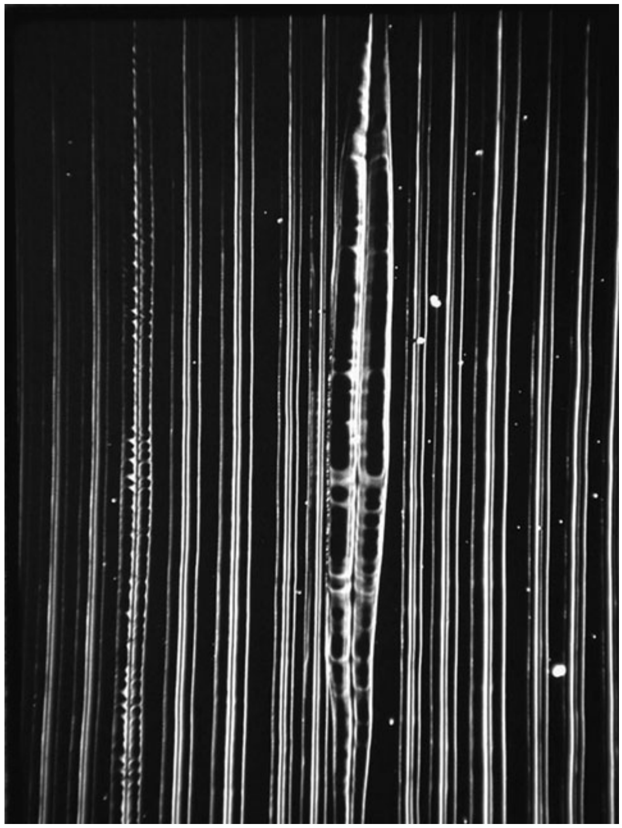


Figure 7.8
Lateral excursions aren't the only source of intercuts. This figure shows a large low-frequency vertical excursion, caused by out-of-phase information, which has been encroached upon by its neighbor during the next revolution. The wall of the later revolution is compromised down to about 0.5 mil. This is not severe enough to cause mistracking; however, some distortion will be heard from the deformity. Since this type of problem arises exclusively from out-of-phase low-frequency information that would be acoustically cancelled upon playback anyway, mono summing is generally performed at low frequencies to eliminate such large vertical excursions.

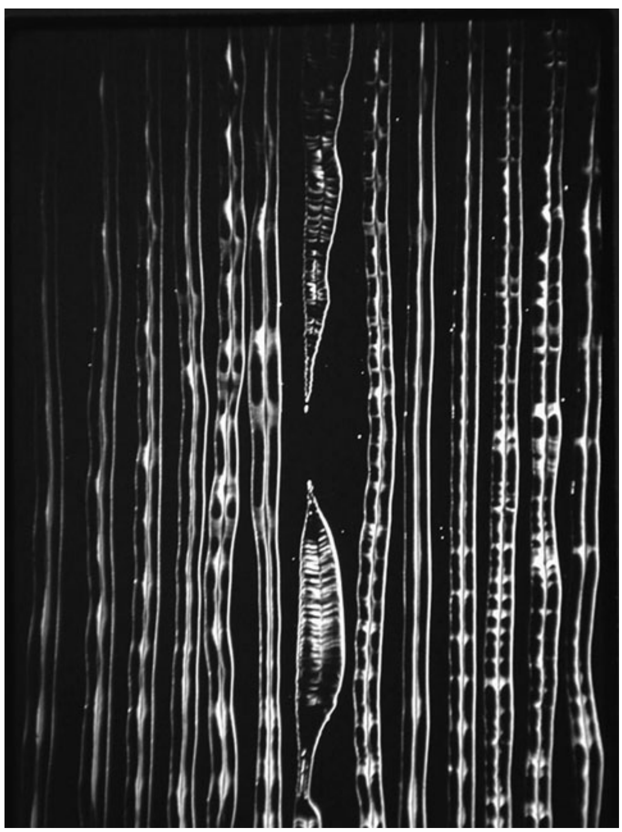


Figure 7.9
Large vertical excursions can cause problems not only by carving out deep, and consequently wide, swaths that result in intercuts, but by causing the cutting stylus to literally lift right off the disk surface for the other half of the waveform. Obviously, a lift such as this would inevitably cause a record to skip and is always unacceptable.

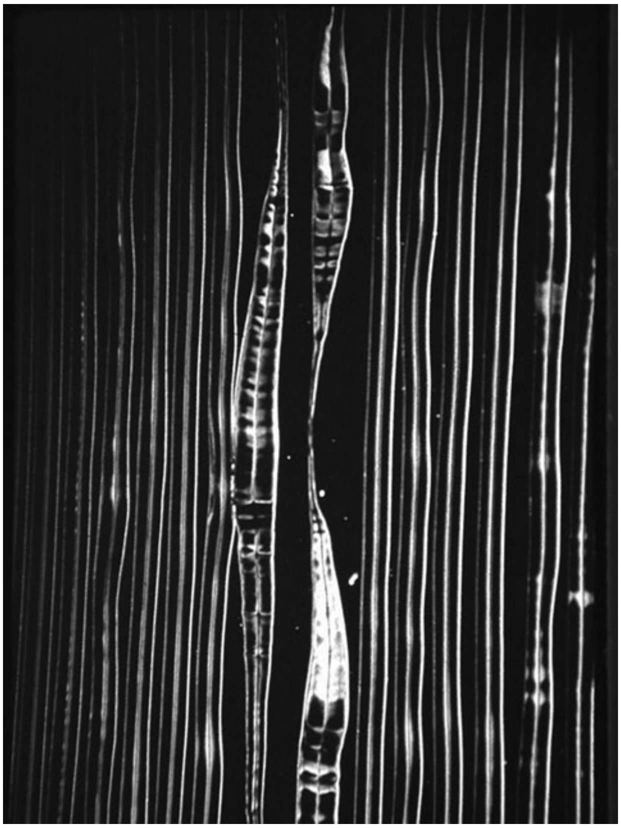


Figure 7.10
Here a near lift (only a tenth of a mil remains of the groove walls) is accompanied on the following revolutions by lateral intercut. The deformity along the inside wall of the new groove as the outside wall encounters the previous revolution is clearly visible opposite the breach. This will result in audible distortion. The mastering engineer has several tools at his disposal to solve problems such as these. Among them are increasing groove pitch and/or depth, lowering the overall level at which the record is cut, reducing low-frequency information, summing low frequencies at a higher crossover point, or adding external processing, such as a peak limiter. Each of these can be used alone or in combination to achieve a satisfactory master, but none can be employed without exacting a price.

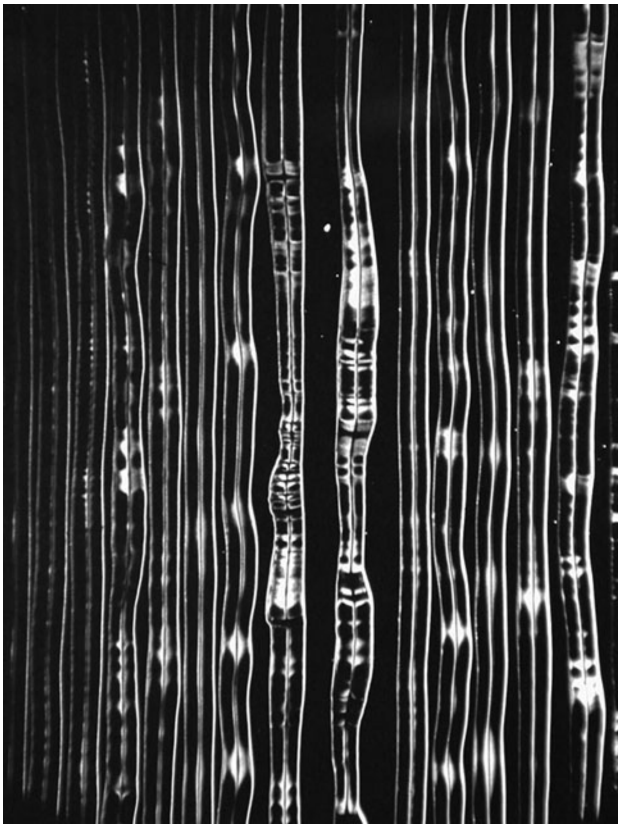


Figure 7.11
Here is the same audio viewed in Figure 7.10, only after processing. In this case a limiter was employed to reduce dynamic range (the surrounding material is noticeably louder as well) and rein in the peaks, which were causing intercuts and lifts. This section is cut more deeply, averaging perhaps 3 to 4 mils instead of the more common 2 mils, in order to give vertical excursions plenty of breathing room. Pitch, too, has had to be increased overall in order to accommodate the slightly wider groove, despite the reduced need for dramatic dynamic increases in pitch due to the reduction of peaks by the limiter.

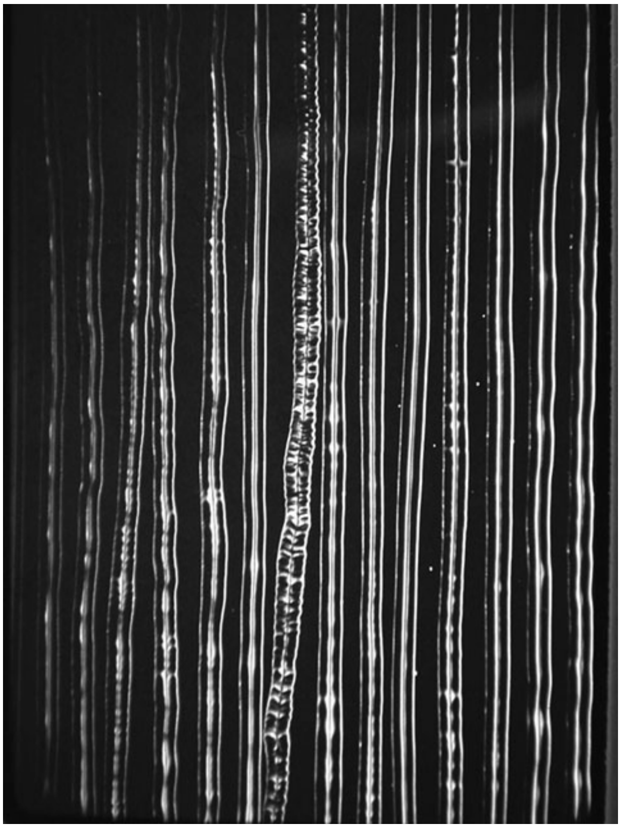


Figure 7.12
Among the truisms of disk cutting: High-frequency information suffers terribly as the groove winds closer to the inner diameter. Here is HF-rich program material near the outer diameter of the disk.

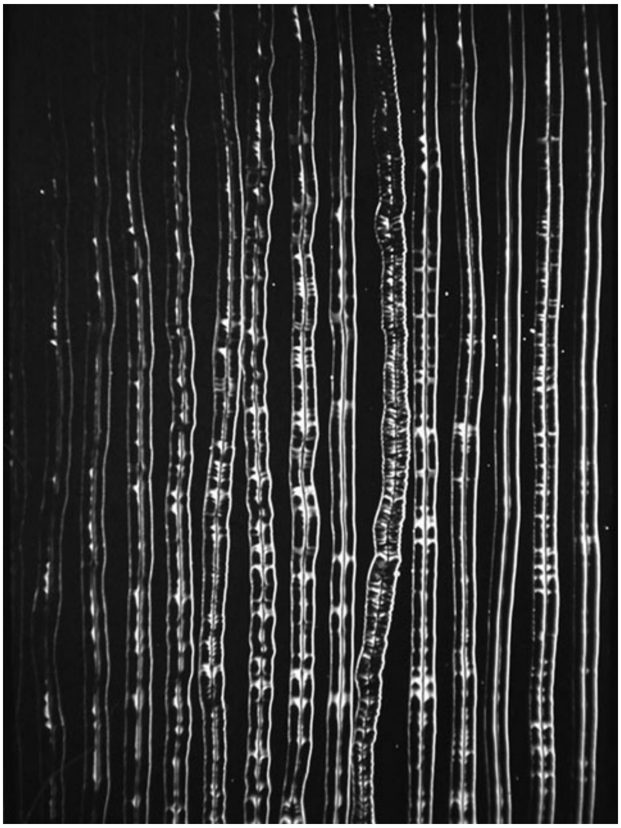


Figure 7.13
Here is the same audio information as in Figure 7.8, only nearer the inside diameter of the disk.

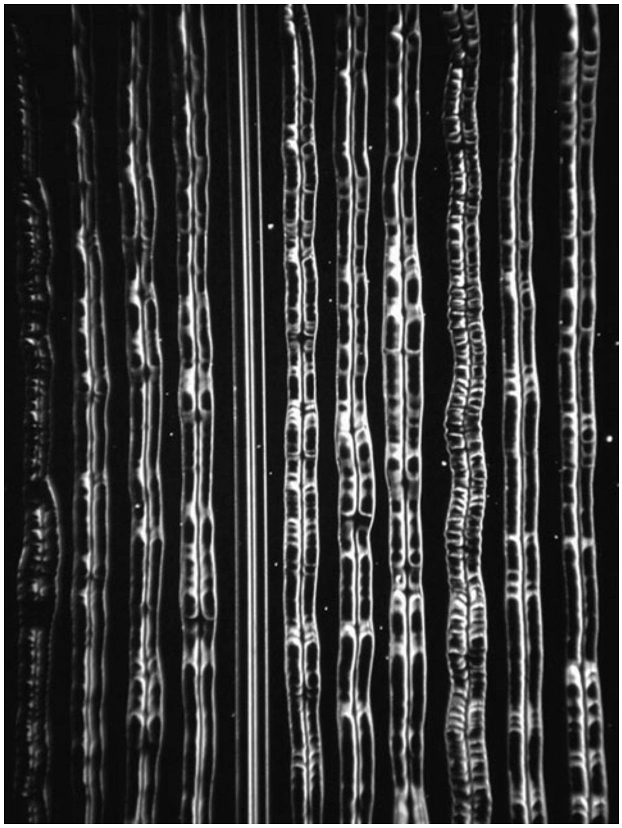


Figure 7.14
The ideal: normal, healthy-looking program audio.

The Vinyl Signal Chain

Although the signal chain for vinyl is similar to that of CD, there are some important distinctions and unique pieces involved. Let's look at the chain from the master lacquer (the record that we cut to send to the pressing plant) on back.

THE MASTER LACQUER

The master lacquer is the record that we cut to send to the pressing plant. It consists of a mirror-smooth substrate of aluminum coated with cellulose nitrate (a distant cousin to nitroglycerine), along with some resins and pigments to keep it soft and help with visual inspection. The lacquer is extremely soft as compared to the finished record and can never be played after it is cut. In order to audition the mastering job before a lacquer is cut, a reference disk called a *ref* or *acetate* is made. Because this is made of the same soft material as on the master lacquer, it can only be played five or six times (at most) before the quality has been significantly degraded. There is a separate master lacquer for each side of the record. The lacquer is always larger than the final record (a 12" record has a 14" lacquer), so repeated handling does not damage the grooves.

THE CUTTING STYLUS AND CUTTER HEAD

The cutting stylus, which is made of sapphire, sits inside the cutter head, which consists of several large drive coils. The drive coils are powered by a set of very high-powered (typically 1,000- to 3,500-watt) amplifiers. The cutting stylus is heated for an easier and quieter cut.

THE LATHE

The lathe contains a precision turntable and the carriage that holds the cutter head assembly, as well as a microscope to inspect the grooves and adjustments that determine the number of grooves and the depth of cut. No lathes are currently being manufactured, but models by Scully and Neumann were once among the most desirable (see Figure 7.15).

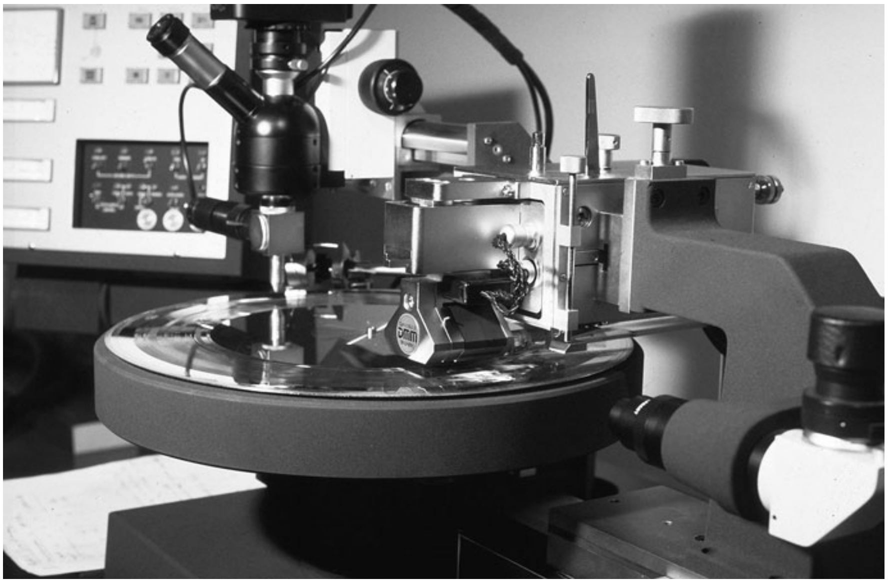


Figure 7.15
Neumann VMS-80 with SX 84 cutter head from 1984. (Image courtesy of Neumann.)

DAVID CHEPPA: *We've actually developed it quite a lot. In the old days, way, way back in the '50s, the first cutting systems weren't very powerful. They only had maybe 10 or 12 watts of power. Then in the '60s Neumann developed a system that brought it up to about 75 watts per channel, which was considered pretty cool. Then in the '70s, the high-powered cutting systems came into being, which were about 500 watts. That was pretty much it for a while. I mean, it made no sense beyond that because the cutter heads really weren't designed to handle that kind of power anyway. Even the last cutting system that came off the line in about 1990 at Neumann in Berlin hadn't really changed other than it had newer panels and prettier electronics. It wasn't really a big difference.*

In the physical world with sound systems, all the energy is in the low end. But in cutting, it's the exact opposite. All of the energy is in the upper spectrum, so everything from about 5,000 cycles up begins to require a great amount of energy. This is why our cutting systems are so powerful. One lathe has 3,600 watts of power, and our least powerful one is about 2,200 watts. It's

devastating if something goes wrong at that power. If I get a master that's raw and hasn't been handled at all and there is something that just tweaks out of nowhere, it can take the cutter head out.

THE MASTERING CONSOLE

The mastering console (see Figure 7.16) for a disk system is equal to that used today for mastering in sound quality and short signal path, but that's where the similarity ends. Because of the unique requirements of cutting a disk and the manual nature of the task (thanks to the lack of computerized gear at the time), there are several features found on this type of desk that have fallen by the wayside in the modern era of mastering.

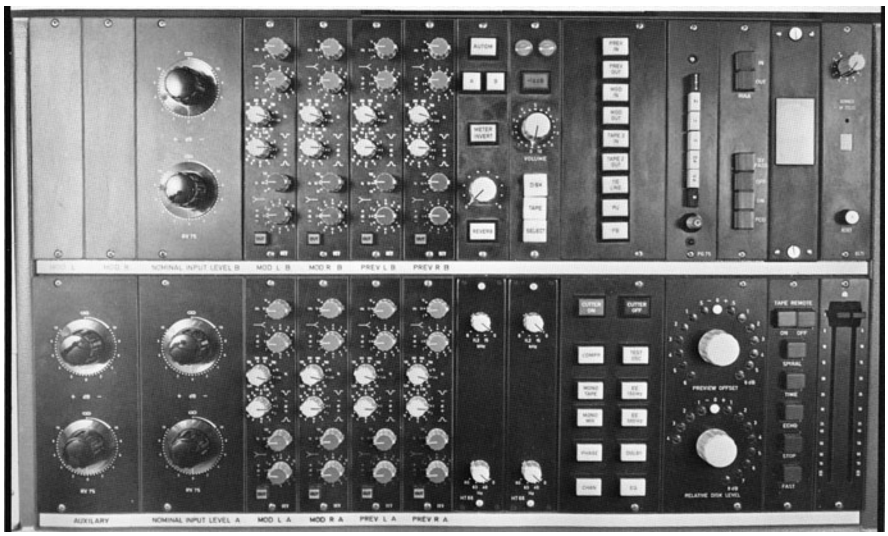


Figure 7.16
A Neumann SP-75 vinyl mastering console.

The Preview System

Chief among those features is the preview system, which is an additional monitor path made necessary by the volatile nature of cutting a disk. Here's the problem: Disk cutting is essentially a non-stop operation. Once you start to cut, you must make all your changes on the fly, without stopping until the end of the side. If a portion of the program had excessive bass information, a loud peak, or something out of phase, the cutter head would cut right through the lacquer to the aluminum substrate. Not only would this destroy the lacquer, but maybe an expensive stylus as well. Hence the need for the mastering engineer to hear the problem and make the necessary adjustments before any harm came to the disk.

Enter the preview system. Essentially, the program going to the disk was delayed. Since digital delays weren't invented yet, an ingenious dedicated mastering tape machine with two separate head stacks (program and preview) and an extended tape path (see Figures 7.17 and 7.18) was used. This gave the mastering engineer enough time to make the necessary adjustments before any damage was done to the disk or system.

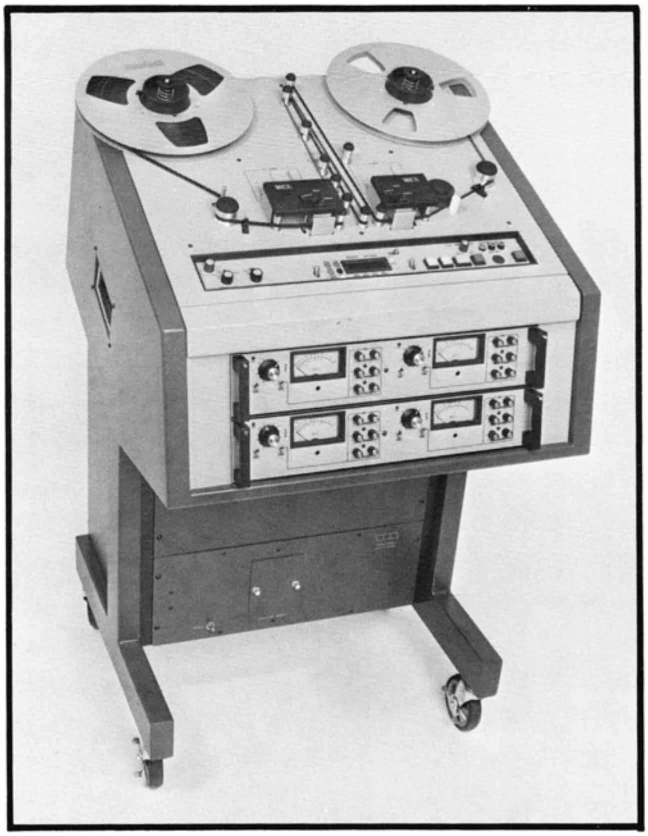


Figure 7.17
MCI tape machine with preview head.

Figure 7.18
Studer tape machine with preview head.

Equalization

Since a disk had to be cut on the fly and computer automation was still years away, a system had to be created in order to make EQ adjustments from song to song quickly, easily, and—most necessarily—manually. This was accomplished by having two of each unit and having the controls of each stepped so that adjustments could be repeatable.

The mastering engineer would then run down all the songs of a side (one side of the LP) and mark down the EQ settings required. Then, as the first song was being cut through the A equalizer, he would preset the B equalizer. As song 2 was playing through the B equalizer, he would preset equalizer A for song 3, and so on (refer to Figure 7.16).

Although this method was crude, it was effective. Naturally, today it's much easier now that all EQ and compression presets can be recalled with only the touch of a button.

The Elliptical Equalizer

One of the more interesting relics of the record days is the *elliptical equalizer* or *low-frequency crossover*. What this unit does is move all low frequencies below a preset frequency (usually 250, 150, 70, and 30 Hz) to the center. This is done to stop excessive lateral movement of the cutting stylus because of excessive low-frequency energy on one side only or excessive out-of-phase material. Obviously, use of this device could negatively affect the sound of a record, so it must be used judiciously.

How Records Are Pressed

Pressing records is such a primitive process by today's standards that it's pretty amazing that they sound as good as they do. This is a multi-step operation that's virtually entirely mechanical and manual, with a host of areas that could influence the end product in a mostly negative way.

STEP 1

The master lacquer is used as the first of several metal molds from which the plastic records are pressed. The lacquer is first coated with a layer of tin and silver nitrate, then dropped in a nickel sulfamate bath and electroplated. The lacquer is removed from the bath, and the nickel coating is peeled away. The separated nickel is what's known as the *metal master* and is a negative of the lacquer.

STEP 2

The metal master is dropped back into the nickel bath and electroplated again. The resultant separated metal part is known as the *mother* and is a positive copy that can be played since it has grooves (although it won't be because to do so would destroy the disc).

STEP 3

The mother is dropped back into the nickel bath and electroplated again. The resultant separated metal part is known as the *stamper* and is a negative copy that is bolted into the record presser to actually stamp out the plastic records.

It should be noted that, just like tape, each resultant copy is a generation down and will result in 6 dB worse signal-to-noise ratio. Also, great care must be used when peeling off the electroplating, since any material left behind will result in a pop or click on the finished product.

STEP 4

The vinyl used to make records actually comes in a granulated form called *vinylite* and isn't black, but honey colored. Before being pressed, it is heated into the form of modeling clay and colored with pigment. At this point it is known as a *biscuit.* The biscuit is then placed in the press, which resembles a large waffle iron and is heated to about 300 degrees. Temperature is important because if the press if too hot, then the record will warp; if it is too cold, then the noise will increase. After pressing, excess vinyl is trimmed with a hot knife, and the records are put on a spindle to cool at room temperature.

All of these metal parts wear out. A stamper will go dull after about 70,000 pressings. Because of that, several sets of metal parts would have to be made for a large order, and in the case of a large-selling record, even several lacquers.

For some nice lathe pictures, go to http://www.aardvarkmastering.com/history.htm.

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CHAPTER 8

Mastering for Internet Distribution

Encoding an MP3 of your mix may seem easy, but to make it sound great requires a bit of thought, some knowledge, and some experimentation. The idea is to encode the smallest file with the highest quality, which is, of course, the tricky part. Here are some tips to get you started in the right direction so you won't have to try every possible parameter combination. Remember, though, that the settings that might work on one particular song or type of music might not work on another.

The Source File

Lossy coding, such as MP3 (check out Chapter 12, "Internet Delivery Formats," for more info), makes the quality of the master mix *more* of an issue because high-quality audio will be damaged much less by this type of encoding than low-quality audio will. Therefore, it's vitally important that you start with the best audio quality (the highest sample rate and the most bits) possible.

It's also important to listen to your encode and perhaps even try a number of different parameter settings before settling on the final product. Listen to the encode, A/B it to the original, and make any additional changes you feel necessary. Sometimes a big, thick wall of sound encodes terribly, and you need to ease back on the compression and limiting of the source track. Other times, heavy compression can make it through better than a mix with more dynamics. There are a few predictions one can make after doing it for a while, but you can never be certain, so listening and adjusting is the only sure way.

MP3 ENCODING TIPS

Here are some things to consider if your mix is intended for MP3 encoding:

- Start with the highest-quality audio file possible.
- Filter out the top end at whatever frequency works best (judge by ear). MP3 has the most difficulty with high frequencies—cutting them out liberates lots of bits (literally) for encoding the lower and mid frequencies. You trade some top end for better quality in the rest of the spectrum.
- A real busy mix can lose punch after encoding. Sparse mixes, such as acoustic jazz trios, seem to retain more of the original audio oomph.
- Make sure your level is reasonably hot. Use the "tips for hot level" (refer to Chapter 4) or even normalize if you must, but it's far better to record at a good level in the first place.
- Don't squander bandwidth. Your encode might actually sound better at 32 kHz than at 44.1 kHz because the encoding algorithm can concentrate on the more critical midrange.
- Don't squash everything with a compressor/limiter. Leave some dynamic range so the encoding algorithm has something to look at.
- Use multi-band compression (such as a TC Electronic Finalizer) or other dynamic spectral effects very sparingly. They just confuse the encoding algorithm.
- Set your encoder for maximum quality, which allows it to process for best results. It takes longer, but it's worth it.
- Remember, MP3 encoding almost always results in the post-encoded material being slightly hotter than the original material. Limit the output of the material intended for MP3 to –1 dB, instead of the commonly used –.1 or –.2 dB, so you don't get digital overs.

The Encoder

Unfortunately, all MP3 encoders are not created equal, and therefore they don't provide the same quality output, so using a good encoder is the biggest advantage you can give yourself.

An MP3 encoder to consider is LAME, which is an open-source application. LAME is an acronym for *LAME Ain't an MP3 Encoder*, although the current version really is a stand-alone encoder. The consensus (as of 2007) seems to be that LAME produces the highest-quality MP3 files for average bit rates of 128 kbps and higher. Another good MP3 encoder is the one found in iTunes.

Bit Rate

Regardless of the encoder, there's really only one parameter that matters most in determining the quality of the encode, and that's bit rate, which is the number of bits of encoded data that are used to represent each second of audio. Lossy encoders like MP3 provide a number of different options for their bit rate. Typically the rates chosen are between 128 and 320 kilobits per second. By contrast, uncompressed audio as stored on a compact disc has a bit rate of about 1400 kbps.

MP3 files encoded with a lower bit rate result in smaller files and therefore faster downloads, but they generally play back at a lower quality. With a bit rate too low, compression artifacts (sounds that were not present in the original recording) may appear in the reproduction. A good demonstration of compression artifacts is provided by the sound of applause, which is hard to data-compress because it is random. As a result, the failings of an encoder are more obvious and become audible as a slight ringing.

Conversely, a high bit rate encode will almost always produce a better sounding file, but also will result in a larger file, which may take an unacceptable amount of time to download.

BIT RATE SETTINGS

For average signals with good encoders, many listeners consider a bit rate of 128 kbps, providing a compression ratio of approximately 11:1, to be near enough to compact disc quality. However, listening tests show that with a bit of practice, many listeners can reliably distinguish 128 kbps MP3s from CD originals. When that happens, many times they reconsider and then deem the 128 kbps MP3 audio to be of unacceptably low quality. Yet other listeners, and the same listeners in other environments (such as in a noisy moving vehicle or at a party), will consider the quality quite acceptable.

- **128 kbps.** This is the lowest acceptable bit rate, but may have marginal quality depending upon the encoder. This results in some artifacts but small file sizes.

- **160 kbps.** This is the lowest bit rate considered usable for a high-quality file.

- **320 kbps.** This is the highest quality with a large file size, but it may be indistinguishable from CD.

CONSTANT VERSUS AVERAGE VERSUS VARIABLE BIT RATE

There are three modes coupled to bit rate that have a bearing on the final sound quality of the encode.

- **Variable Bit Rate mode (VBR).** This maintains a constant quality while raising and lowering the bit rate depending upon how complex the program is. Size is less predictable than with ABR (see below), but the quality is usually better.

- **Average Bit Rate mode (ABR).** This varies the bit rate around a specified target bit rate.

- **Constant Bit Rate mode (CBR).** This maintains a steady bit rate regardless of the complexity of the program. CBR mode usually provides the lowest-quality encode, but the file size is very predictable.

At a given bit rate range, VBR will provide higher quality than ABR, which will provide higher quality than CBR. The exception to this is when you choose the highest possible bit rate of 320 kbps where, depending upon the encoder, the mode may have little bearing on the final sound quality.

Other Settings

There are some additional parameter settings that can have a huge influence on the quality of the final encode. These include:

- **Mid-Side Joint Stereo (sometimes called MS Joint Stereo).** This encodes all of the common audio on one channel and the difference audio (stereo minus the mono information) on the other channel. This is intended for low bit-rate material and to retain surround information from a surround mix source. It is not needed or desired for stereo source files. *Do not select this under normal circumstances.*

- **Intensity Joint Stereo.** Again intended for lower bit rates, Intensity Joint Stereo combines the left and right channels by saving some frequencies as mono and placing them in the stereo field based on the intensity of the sound. *This should not be used if the stereo audio contains surround-encoded material.*

- **Stereo Narrowing.** Again intended for lower bit rates, this allows narrowing of the stereo signal to increase overall sound quality.

It's better not to check any of the above parameters when encoding stereo files that originate at 16 bits or above. With these disabled, the encoding will remain in true stereo, with all of the information from the original left channel going to the left side and the same for the right channel.

For Best MP3 Encodes

- Don't hypercompress the source master.
- Cut some of the high frequencies.
- Use Variable Bit Rate mode.
- Turn off Mid-Side Joint Stereo, Intensity Joint Stereo, and Stereo Narrowing.
- Try not to use a bit rate below 160 kbps (higher is better).
- Set the output to –1 dB since encodes are hotter.

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CHAPTER 9

Mastering in Surround

With surround sound production now more or less commonplace, producers now find they need the same finishing touches of mastering a surround mix that they've long been accustomed to in stereo. As a result, mastering facilities worldwide have upgraded to the brave new world of multichannel. Perhaps even more than in recording and mixing, mastering in this environment requires greater thought, planning, and skill than other audio facilities face. In surround mastering, it's not just a question of adding four channels of additional equipment and carrying on as before. The question really is, will the client expect other services as well?

Here are some of the concerns faced by the mastering engineer contemplating surround sound.

First a Bit of History

Surround sound in one form or another has actually been with us for more than 50 years. Film has always used the three-channel "curtain of sound" developed by Bell Labs in the early 1930s. This was because it was discovered that a center channel provided the significant benefits of anchoring the center by eliminating "phantom" images (in stereo the center images shift as you move around the room) and better frequency response matching across the sound field.

The addition of a rear effects channel to the front three channels dates as far back as 1941, with the Fantasound four-channel system utilized by Disney for the film *Fantasia*, and in the 1950s, with Fox's CinemaScope. Still, the rear channel didn't come into widespread use until the 1960s, when Dolby Stereo became the *de facto* surround standard. This popular

film format uses four channels (left, center, right, and a mono surround—sometimes called *LCRS*) and is encoded onto two tracks. Almost all major television shows and theatrical releases are presented in Dolby Stereo because it has the added advantage of playing back properly in stereo or mono if no decoder is present.

With the advent of digital delivery formats capable of supplying more channels in the 1980s, the number of surround channels was increased to two, and the low-frequency effects channel was added to make up the six-channel 5.1, which soon became the modern standard for most films, music, and DTV. The *Star Wars* prequel *Episode I—The Phantom Menace* introduced the Dolby Digital Surround EX 6.1 format (DTS soon followed with their ES version), in which a center rear channel is used. And Sony Dynamic Digital Sound, or SDDS, offers a 7.1 with two additional screen channels called *Left Center* and *Right Center*.

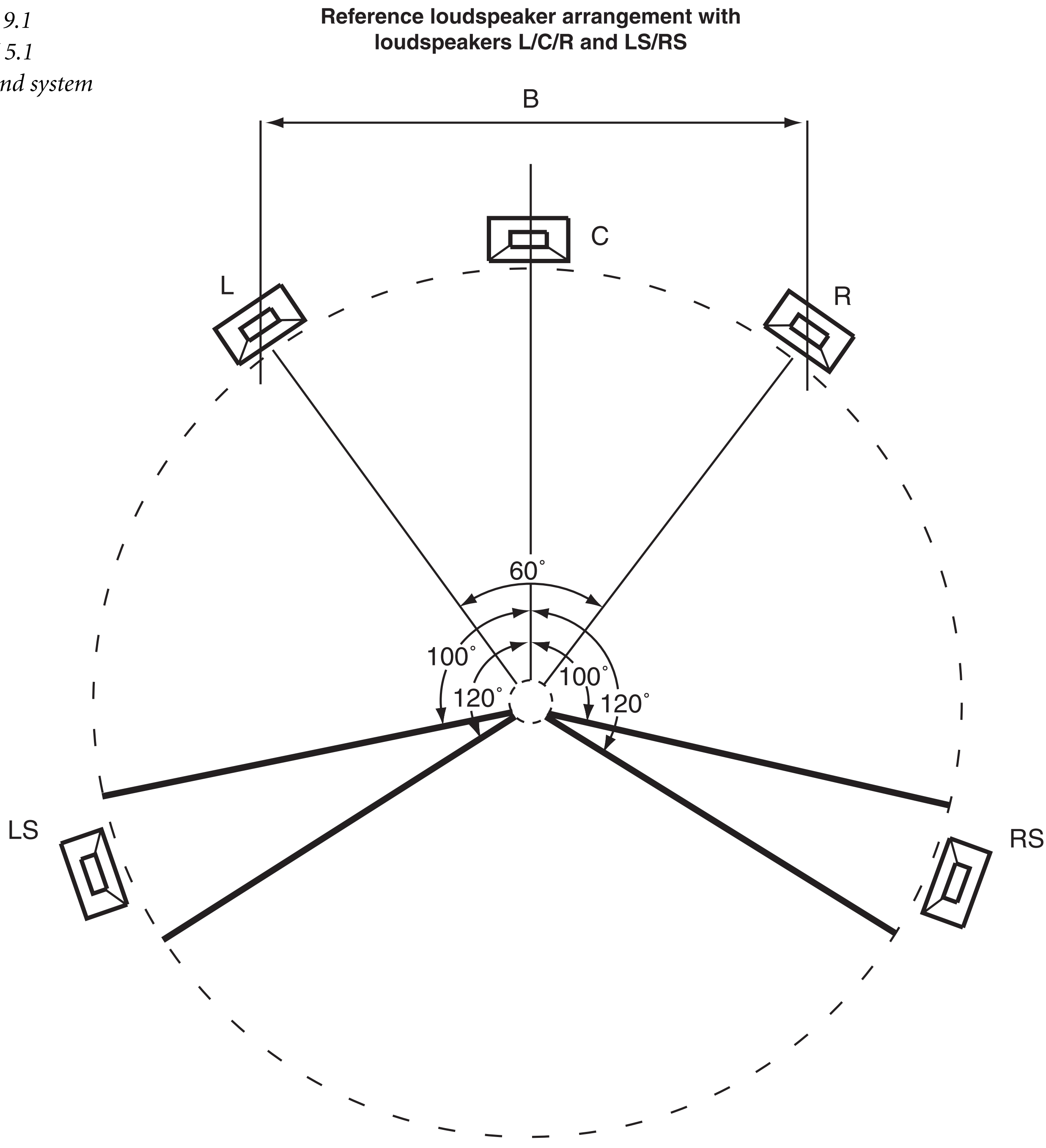
And of course, there's always Quad from the 1970s, the music industry's attempt at multichannel music that killed itself as a result of two non-compatible competing systems (a preview of the Beta versus VHS war) and a poor psychoacoustic rendering that suffered from an extremely small sweet spot.

Types of Surround Sound

The format known as 5.1 is the mostly widely used surround format today, being used in motion pictures, music, and digital television. The format consists of six discrete speaker sources—three across the front (left, center, and right) and two in the rear (left surround, right surround), plus a subwoofer known as the *low-frequency effects* channel, or LFE, which is the *.1* of the *5.1* (see Figure 9.1). This is the same configuration that you hear in most movie theatres because 5.1 is the speaker specification used not only by THX, but also by popular motion-picture release formats such as Dolby Digital and DTS.

Figure 9.1 shows what's known as *ITU Specification 775*, which was an early attempt to standardize the setup of surround speaker systems. This setup, though still frequently used, was used primarily for listening to classical music, rather than anything modern. Although it still can work for rock, R&B, and so on, most surround mixers have settled on a setup of equidistant speakers almost in a triangle. In practice, the location of the speakers (even though not ideal) is very forgiving as long as the system is calibrated properly.

Figure 9.1 A ITU 5.1 surround system setup.



B: Loudspeaker Base Width

Loudspeaker	Horizontal Angle from Center (degrees)	Height (m)	Inclination (degrees)
C	0	1.2	0
L,R	30	1.2	0
LS, RS	100...120	≥1.2	0...15 down

The LFE Channel

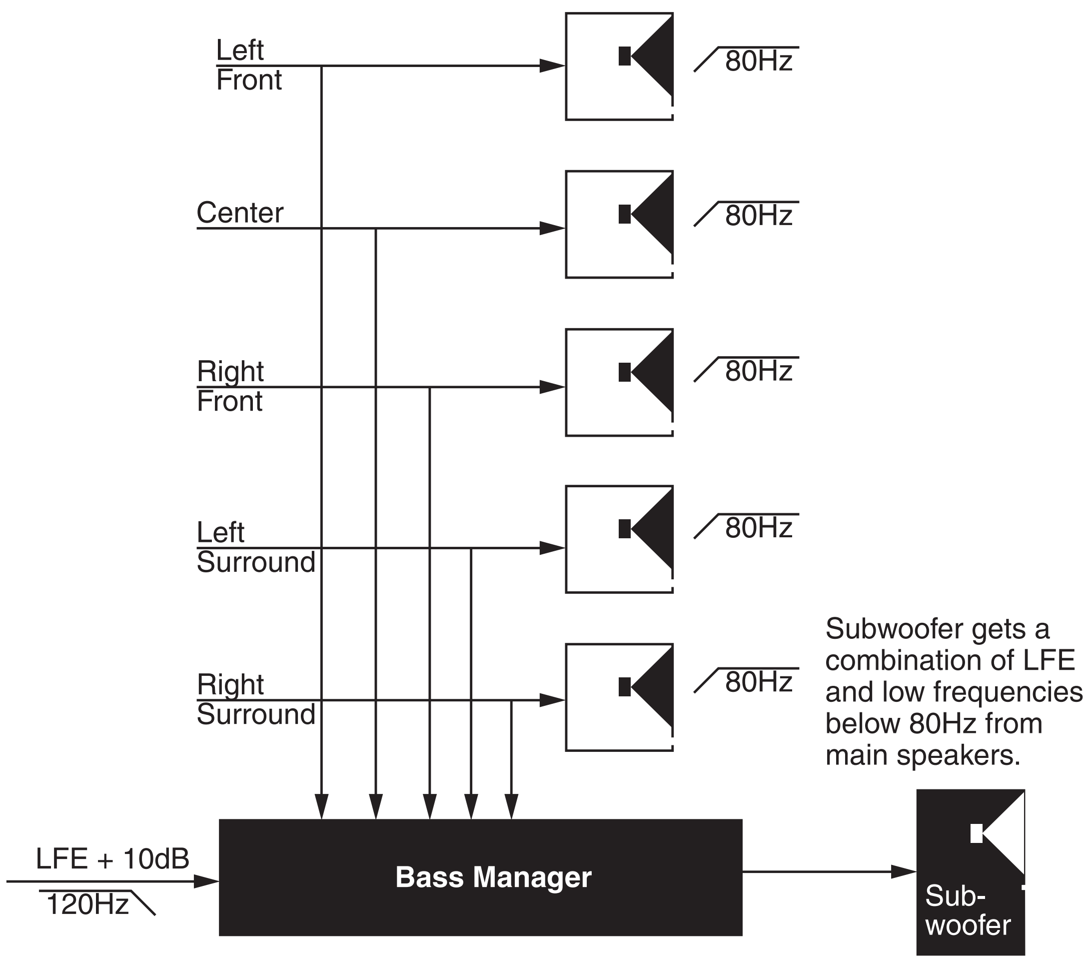
LFE is sometimes referred to in film-production circles as the *Boom channel* because that's what it's there for—to enhance the low frequencies of a film so you get the extra boom out of an earthquake, plane crash, explosion, or other such dramatic scene requiring lots of low frequencies.

The LFE, which has a frequency response from about 25 Hz to 120 Hz, is unique in that it has an additional 10 dB of headroom built into it. This is needed to accommodate the extra power required to reproduce the low-frequency content without distortion.

Bass Management

The *bass manager* (sometimes called *bass redirection*) is a circuit that takes all the frequencies below 80 Hz from the main channels (according to the Dolby spec) and the signal from the LFE channel and mixes them together into the subwoofer (see Figure 9.2). The reason why this is done is to make

Figure 9.2
Bass management.



use of the subwoofer for more than the occasional low-frequency effect, since it's in the system already. This enables the effective response of the system to be lowered to about 25 Hz.

Because the overwhelming majority of consumer home-theater systems (especially the average low-end ones) contain a bass management circuit, there's a school of thought that says you should use one in the studio in order to hear things the way the people at home hear them. Otherwise, consumers may actually be hearing things (such as unwanted rumbles) that you can't hear because the bass manager gives a low-frequency extension below that of the vast majority of studio monitors. That being said, it's not uncommon for a bass management circuit not to be used during mixing and mastering, or to just be occasionally switched in and out for a quick check.

Other Types of Surround

There are many other widely used surround formats. Three-channel (stereo front speakers with a mono surround); four-channel (three front speakers with a mono surround), such as Dolby Pro Logic; five-channel (three front speakers with a stereo surround but no LFE channel), such as Dolby Pro Logic II; and seven-channel (the Sony SDDS format with five front speakers) all abound. It's important to note that very few A/V receivers are able to reproduce 7.1 (they automatically downmix to 5.1), and only Blu-ray and HD-DVD discs are even capable of utilizing the format. What's more, it still hasn't been determined whether the extra two speakers will be used on the sides or in the front, as in SDDS.

There are other non-standard formats that use as many as 10 channels for height and extra rear and side channels as well. Dolby Digital EX and DTS-ES take film sound to a new level by adding a center rear channel, something that film mixers have been asking for more and more. And many amusement rides and Las Vegas showrooms now use as many as 30 channels to enhance the surround experience.

Table 9.1 lists a number of different surround types.

Table 9.1 Different Surround Types

Codec	Max # of Channels	Delivery Method
Dolby Surround	4 (L, C, R, S)	Cinema
Dolby Pro Logic	4	Broadcast
Dolby Pro Logic II	5 (L, C, R, Ls, Rs)	Broadcast, Game Consoles
Dolby Pro Logic IIx	6.1 (L, C, R, Ls, Cs, Rs, LFE)	Broadcast
Dolby Digital (AC-3)	5.1 (L, C, R, Ls, Rs, LFE)	Cinema, DVD, HDTV
Dolby Digital EX	6.1	Cinema, DVD
Dolby Digital Plus (DD+)	7.1(L, C, R, Ls, Rs, Lr, Rr, LFE)	HD-DVD, Blu-ray
Dolby E	7.1	Production Only
Dolby TrueHD	7.1	HD-DVD, Blu-ray
Meridian Lossless Packing (MLP)	5.1	DVD-A, HD-DVD, Blu-ray
SRS	6.1	Broadcast
DTS Digital Surround	5.1	Cinema, DVD
DTS Digital Surround ES	6.1	Cinema, DVD
DTS Digital Surround 96/24	5.1	DVD, HD-DVD, Blu-ray
DTS-HD	7.1	HD-DVD, Blu-ray
DTS-HD Master Audio	7.1	HD-DVD, Blu-ray
SDDS (Sony Dynamic Digital Sound)	7.1 (L, C, R, Lc, Rc, Lr, Rr, LFE)	Cinema

Legend: L = Left Front, R = Right Front, C = Center Front, Ls = Left Surround, Rs = Right Surround, Cs = Center Surround, Lr = Left Rear, Rr = Right Rear, LFE = Low Frequency Effects (subwoofer), Lc = Left Center Front, Rc = Right Center Front

The Differences between Surround and Stereo

When you listen to a good surround-sound mix, you'll notice quite a few differences (some might say improvements) over stereo:

- The sonic clarity is enhanced because the center channel anchors the sound and eliminates any "phantom" image shifts that we take for granted in stereo.

- There is no sweet spot per se. Actually, the whole room becomes a sweet spot in that you can move around freely and never lose the sense of clarity, dimension, and spatial continuity. One listener described it perfectly as an “audio sculpture” in that, just like when you walk around a piece of artwork and get a different perspective of the art, when you walk around the 5.1 room you just get a different perspective of the mix. You might get closer to the guitar player, for instance, if you walk to the left of the room. Walk to the right, and you’re closer to the piano. Indeed, you don’t have to even be in the speaker field to get a sense of the depth of the mix. Even people sitting outside the soundscape often describe an enhanced experience.

- Speaker placement is very forgiving. Yes, there are standards for placement, but these tend to be very non-critical. The sense of spaciousness remains the same regardless of how haphazardly the speakers are distributed around the room. In fact, stereo is far more critical placement-wise than surround sound.

Differences between Surround Mixes for Picture and for Music

Normally in the theater, all of the primary sound information comes from the front speakers, and the surround speakers are utilized only for ambience info, in order to keep your attention on the screen. The LFE is intended to be used just for special effects such as explosions and earthquakes, and it is therefore used infrequently. One of the reasons that the surround speakers don’t contain more source information is a phenomenon known as the *exit-sign effect*, which means that your attention is drawn away from the screen to the exit sign when the information from the surrounds is too loud.

But music-only surround sound has no screen to focus on and therefore no exit-sign effect to worry about. Take away the screen, and it’s now possible to utilize the surround speakers for more creative purposes.

Different Perspectives: Audience versus Onstage

There are two schools of thought about how surround sound for music should be mixed. The *audience* or *classical* perspective puts the music in the front speakers and the hall ambience in the surrounds, just as if you were sitting in the audience of a club or concert hall. This method may not utilize the LFE channel at all and is meant to reproduce an audience perspective of the musical experience.

The second is the *onstage* perspective. In this case the band is spread all over the room via the five main speakers, and that puts the listener in the center of the band and envelops him with sound. This method usually results in a much more dramatic soundstage that is far larger-sounding than the stereo that we're used to. This may not be as authentic a soundscape as some music (any kind of live music where the listeners' perspective is from the audience) might require, however.

The Center Channel

In film mixing, the center channel is used primarily for dialogue so the listener doesn't get distracted by sonic movement. In music, however, its use prompts debate among mixers.

NO CENTER CHANNEL

Many veteran engineers who have mixed in stereo all their lives have trouble breaking the stereo paradigm to make use of the center channel. These mixers continue to use a phantom center from the left and right front speakers and prefer not use center channel at all.

ISOLATED ELEMENTS IN THE CENTER CHANNEL

Many mixers prefer to use the center channel to isolate certain elements, such as lead vocals, solos, and bass. Although this might work in some cases, many times the isolated elements seem disconnected from the rest of the soundscape.

THE CENTER AS PART OF THE WHOLE

Mixers who use the center channel to its fullest find that it acts to anchor the sound and eliminates any drifting phantom images. In this case, all five speakers have equal importance, with the balance changing the sound elements placed in the soundscape.

THE LFE (SUBWOOFER) CHANNEL

Anything that requires some low-frequency bass extension can be put into the subwoofer via the LFE channel. Many mixers put a little kick and/or bass there if it's used at all. Remember that the frequency response only goes up to 120 Hz, so the definition from the instrument actually comes from the main channels.

Surround Master Media Prep

Surround sound brings a new level of complexity not normally found in stereo. Therefore, it's imperative to indicate as much information about the project as possible. You can avoid many potential problems as long as the master is prepped and the items discussed in the following sections are noted.

These items apply not only to the mastering engineer before sending a project to authoring, but even more so to the mixing engineer before sending the final mixes to mastering. Therefore, it's important for the mastering engineer to communicate their importance to the mixer prior to getting a project.

SLATE THE MASTER

More than ever before, it's important to not only properly document the master tape or disc, but to prep the master to make sure that there are no questions as to the actual track assignments. Even an engineer who has mixed the tracks sometimes has a hard time determining which is the center channel and which is the left surround, so it's quite necessary to take any guesswork out of the process.

The best way to avoid confusion is to go back to the admittedly low-tech but foolproof method of using an audio slate on each channel indicating the channel assignment (such as "Channel One - Left Front," "Channel Six - Right Surround," and so on).

Master Tape Track Assignments

Sooner or later the question of channel assignment on the master recorder (be it tape or hard disc) always arises. What is the correct track assignment? Actually there are several generally accepted channel assignment formats for surround, although the first (see Table 9.2) is fast becoming the *de facto* standard.

Table 9.2 The SMPTE and ITU Standard Channel Assignments

1	2	3	4	5	6
Left Front	Right Front	Center	LFE	Left Surround	Right Surround

A dedicated stereo mix, or Lt/Rt (Left Total/Right Total), or encoded AC3 can be recorded onto Tracks 7 and 8.

This format is the SMPTE and ITU standard, as well as the assignment matrix suggested by Dolby, and transfers easily to the corresponding four audio tracks (L, R, C, LFE) of the most widely used video formats today, such as DigiBeta or D5. It is also the recommended format by Dolby as it is the common pairing of channels in Dolby Digital encoding (although the AC-3 encoder can actually be configured to any track configuration).

The following two assignment methods (see Tables 9.3 and 9.4) are also used, but less and less as the SMPTE/ITU standard becomes more and more widespread.

Table 9.3 Preferred Film Channel Assignment

1	2	3	4	5	6
Left Front	Center	Right Front	Left Surround	Right Surround	LFE

The Table 9.3 configuration is what many film studios use, although it's seen in some music production as well. It seems to make sense in that it's a somewhat visual representation of the way the speakers are laid out, but it falls short when it comes to logical track pairings.

Table 9.4 shows the channel assignments preferred by DTS. Again, the pairings are logical, but the placement is different from the SMPTE/ITU standard. Tracks 7 and 8 usually contain the stereo version of the mix, if one is needed.

Table 9.4 DTS Standard Channel Assignments

1	2	3	4	5	6
Left Front	Right Front	Left Surround	Right Surround	Center	LFE

SDDS (*Sony Dynamic Digital Sound*) is a special case in that it's a 7.1 format. SDDS uses a track assignment that differs from the norm, but again makes sense because it gives you a visual representation of the way that the speakers are laid out (see Table 9.5).

Table 9.5 SDDS Channel Assignments

1	2	3	4	5	6	7	8
Left Front	Left Center	Center	Right Center	Right Front	LFE	Left Surround	Right Surround

There are obviously other assignment permutations that are occasionally used, but all seem to be falling quickly by the wayside as the SMPTE/ITU track assignment method takes hold.

PRINT A TEST TONE

If the master delivery is on tape (most likely a DA-88 format), be sure to print at least 30 seconds of 1-kHz tone at –20 dBFS, which is the SMPTE standard reference level, across all tracks. A 1k tone is a pretty good way to discover whether there are any clock discrepancies since the purity of the signal will suffer as a result of clicks and warbles that might not be heard during the actual program material.

Also keep in mind that any program on tape media should start no earlier than two minutes into the tape, since that's where most errors and dropouts usually occur.

PRINT TIME CODE

If the audio program is intended for DVD in any form, time code is necessary to maintain sync when it is authored. Generally speaking, it's safest to use 29.97-Drop Frame SMPTE on audio-only program because it is the NTSC color television standard. If a music video is later added to the program (which can cause a multitude of additional problems), it's highly likely that the picture will be at that frame rate. Audio that must be synched to existing picture must use the existing picture time-code frame rate, however.

SURROUND-TO-STEREO COMPATIBILITY

Although it's possible to have the surround mix automatically downmixed to stereo by selecting the downmix parameters on the Dolby Digital encoder, the results are often unpredictable and many times unsatisfactory. Because many surround mixes will default to stereo if only two speakers are present (such as when played in the DVD drive of a computer), it's as important to check the surround-to-stereo compatibility as it is to check the stereo-to-mono compatibility.

DOCUMENT THE DETAILS

Once again, you must indicate the following details to avoid confusion later during the authoring process.

- **Is the LFE channel filtered, and at what frequency?** This is important if for no other reason than it's easy to figure out which is the subwoofer channel if the assignment documentation is lost. The LFE should have a low-pass filter that cuts off at 120 Hz.

- **What is the reference level in SPL?** This helps to better approximate what you were hearing if the program should require remastering. Typical reference levels are 85 dB SPL (the film reference) or 79 dB SPL (the television reference).

- **What is the sampling rate?** This helps to avoid any clock or sync issues that may arise during authoring. Depending upon the ultimate distribution media, the sample rate can be any number of standard rates. For instance, take a look at Table 9.6.

Table 9.6 Legal Sampling Rates

DTS Music Disc	44.1 kHz
DVD-Video	48 kHz multichannel, 96 kHz stereo
DVD-Audio	44.1, 48, 88.2, 96, 176.4, or 192 kHz
Blu-ray	44.1, 48, 88.2, 96, 176.4, or 192 kHz
HD-DVD	44.1, 48, 88.2, 96, 176.4, or 192 kHz

- **What is the bit resolution?** Once again, the type of distribution media will determine the bit resolution (see Table 9.7).

Table 9.7 Legal Bit Resolution

DTS Music Disc	20-bit
DVD-Video	16- to 24-bit
DVD-Audio	16- to 24-bit
Blu-ray	16- to 24-bit
HD-DVD	16- to 24-bit

- **What is the time code format?** As stated before, if the audio program is linked to picture or intended for DVD in any form, time code is necessary to maintain sync. The frame rate chosen must be indicated to avoid later confusion.

- **Are the surround channels calibrated equal to the front channels or –3 dB?** In film-style mixing, the surround channels are calibrated 3 dB down from the screen channels. Music-style mixing has the surrounds equal in level to the front speakers.

- **What is the media format and how many pieces are there?** The master elements may be on several pieces of media across several different formats. A warning here about which piece of media contains the audio master can eliminate the confusion of an incomplete authoring job later.

- **How long is the program?** This is necessary because it determines whether data compression must be used during authoring and helps with managing the total bit budget for the entire DVD, Blu-ray, or HD-DVD disc.

CHAPTER 10

Surround Tools

Although there are many similarities between stereo and surround mastering gear, the unique requirements for surround mastering are more than just some additional channels.

Monitoring

To any mastering facility, its monitor resolution is its major selling point. It is the gold standard, second only to its engineers, by which its clients perceive the facility. While the monitors used in mastering have long been largely a personal choice (even more so than in recording studios), more variables than ever lay ahead when choosing a surround system for the mastering studio.

Up until about 2003, a major question facing anyone getting into surround was, "Should the monitor choice be five identical direct radiator-type (front-firing) speakers, or should the surrounds be dipoles (speakers in which the sound emanates from the sides instead of from the front)?" Many of the original surround recordings were orchestral music, and it was felt that dipoles provided a more accurate sound of the hall. Dipoles have fallen by the wayside in favor of the common direct radiator, however, and the type of surround speaker is no longer an issue.

The issues that do come up frequently are whether to use bass management (see the following section) and whether it's beneficial to use a consumer multichannel receiver as a reference.

Bass Management

Bass management is an area of both great importance and great confusion. It's imperative that the mastering engineer not only hear at the highest resolution possible, but also know that what he's hearing will translate correctly to the consumer in the home. Once again, virtually all of the 50 million home surround systems currently employ some sort of bass manager. Therefore, bass management (sometimes also referred to as *bass redirection*) must be properly implemented in the mastering studio in order for low-end compatibility to occur, even if it's only used for the occasional check. If bass management isn't employed, it's entirely possible that the consumer with a high-quality home theater system will hear things in the subwoofer (because of the low-frequency extension of the system) that the mastering engineer cannot.

Test Equipment

With speaker alignment more critical than ever, it is of utmost importance for the mastering facility to have the proper test gear available to keep the system properly adjusted. Gone are the days when a Sonopulse or a Radio Shack SPL meter and some wide-band pink noise kept things merely close enough. A multichannel test disc (such as Tomlinson Holman's Test and Measurement Series, distributed by Hollywood Edge) along with a spectrum analyzer or an Audio Toolbox (see Figure 10.1) is now a must in order to adjust the level of the subwoofer to the required precision, although some of the newer 5.1 speakers systems from JBL (LSR 4000 series) and Genelec (8200 series) are now self-calibrating.

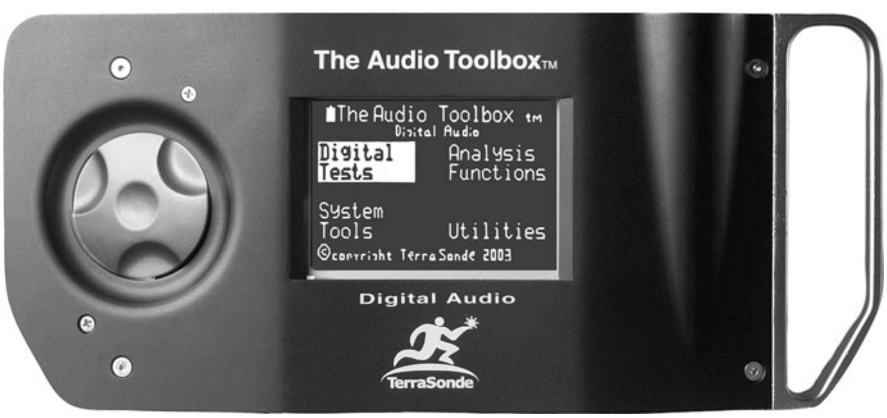


Figure 10.1
TerraSonde Digital Audio Toolbox.
(Image courtesy of TerraSonde.)

The Monitor Controller

While most mastering consoles have always been a somewhat custom item, a surround mastering console requires features that are no trivial matter. Besides the minimum six channels, the major component of the surround console is monitor level control, which must be precisely calibrated to increase or decrease the volume level as needed without disturbing the balance between the main monitors and the subwoofer, or the front speakers and the surrounds. The ability to switch between several surround systems (A/B switching), listen through a decoder, listen to surround formats other than 5.1, as well as perform stereo and mono monitoring, is vital to the final product, and these capabilities must be included as well. Many excellent aftermarket monitor control products are presently available, including the Martinsound MultiMAX, EMM Labs Switchman MKII, and Grace Design m906 (see Figure 10.2).

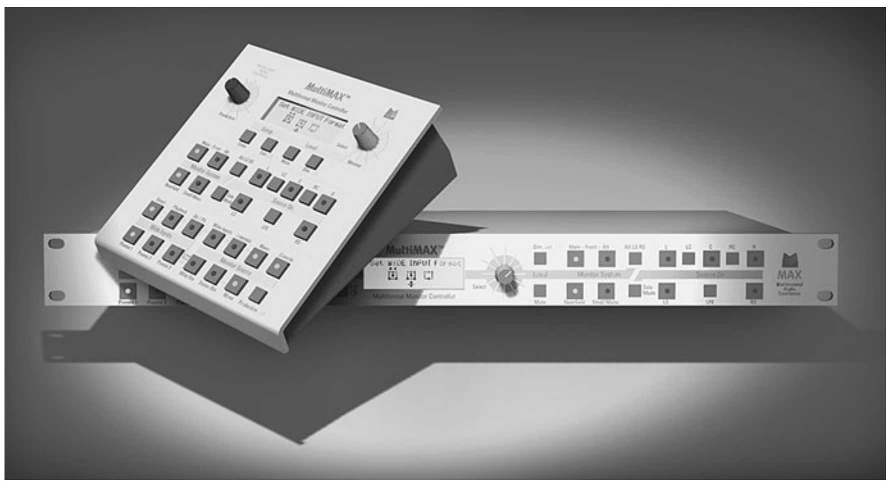


Figure 10.2
Martinsound MultiMAX surround controller.

Converters

Although it's a given that much of the program material will be delivered in the digital domain, that doesn't preclude the need for at least six channels (preferably eight) of high-quality A/D and D/A conversion (refer to Chapter 3). Many items in the mastering engineer's bag of tricks are still analog, and the ability to jump domains must be readily available. Also, some producers mix to 1" or even 2" eight-track analog both for the sound and for archival purposes, making these additional converters an immediate necessity.

Outboard Gear

Not as simple as just adding extra channels, proper ergonomics must accompany any multichannel outboard unit to make its operation fast and easy for the mastering engineer. Compressors and equalizers must have the added capability of not only being ganged for multichannel operation in multiple configurations (two, three, four, five, and six channels), but must also have the ability to have each channel individually tweaked as well (see Figure 10.3).

Figure 10.3
Z-Systems six-channel equalizer and compressor.

Ergonomics of these devices must be extremely user friendly (a highly overused but all too appropriate term) because the mastering engineer by nature does many repeatable operations (such as equalization) very quickly. These operations now increase with the addition of at least four channels. With the many new variables now facing engineers, great pains must be taken to avoid multiple pages and deep menus that slow the process down.

Software Tools

Although hardware for surround mastering was once not only scarce but expensive, there is now a variety of software tools that can accomplish almost any task right in the box (and fairly inexpensively, too). Most DAWs now come with either 5.1 plug-ins or the means to configure the

existing plug-ins for surround. There are some third-party surround plug-ins that are very useful, though.

WAVES 360° SURROUND

This bundle of surround tools has almost everything you need to successfully master 5.1. Included are a surround panner, imager, reverb, compressor, limiter, bass manager, and controller modules. I must admit that I'm partial to this bundle since I consulted on it and wrote parts of the manual. This is also the only package with a separate limiter and compressor, both of which are necessary for a complete master job.

STEREO-TO-5.1 CONVERSION

There are many times when a full surround mix is not possible, and a 5.1 mix must be derived from a stereo program. Believe it or not, there are several plug-ins on the market that can do a pretty good job of this. You still need ears and a little expertise to really get a convincing product, but under the right circumstances, the results can be remarkable.

Two of the best plug-ins for this are TC Electronic's UnWrap and Cycling '74's UpMix. UnWrap (see Figure 10.4) has a lot of parameters to tweak, but if the mix is wide with a lot of center information, you'll get some surprising results (good enough to fool a lot of pros). UpMix, the brainchild of the excellent surround engineer Ron MacLeod (who also produces a set of great effects libraries) has a few more parameters (FoldDown, LFE Generator, and Rotator) that come in handy when extra adjustment is necessary.

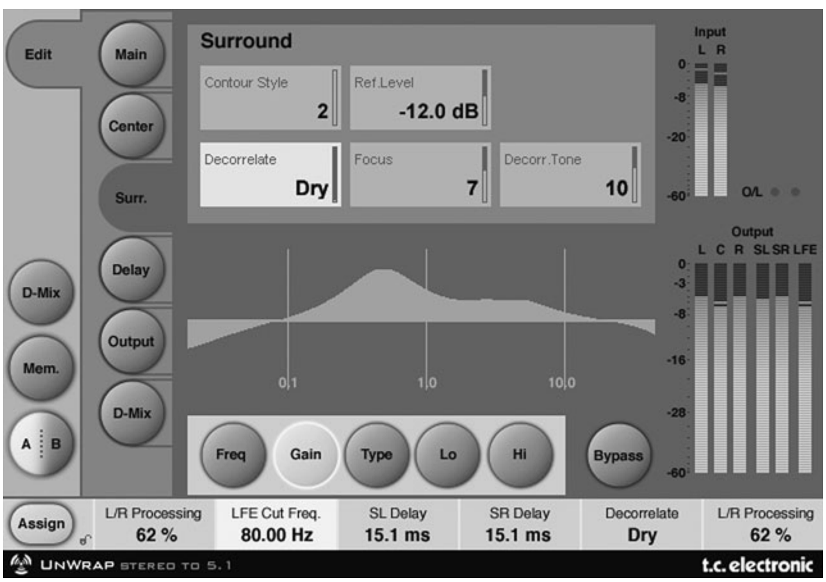


Figure 10.4
TC Electronic UnWrap Stereo-to-5.1 upmix.

96/24 AND BEYOND

With DVD-Audio, Blu-Ray, and HD-DVD discs now a reality, the demand for at least some form of 96-kHz/24-bit—and even 192-kHz/24-bit—audio is growing rapidly. With the increased sample rate and bit depth come the obvious problems of storage and backup, which, although voracious enough in stereo, becomes humongous in 96/24.

Consider this: We all know that a 48/16 stereo minute needs approximately 11.5 MB of storage (actually 11.52 MB). A minute of true 96/24 stereo needs 34.56 MB, and a minute of discrete 5.1 surround at 96/24 requires a whopping 104 MB! This means that a 60-minute program will need 6.24 GB just to get it into the DAW. With the capacity of a basic DVD 5 at 4.32 GB, now it's easy to see why some form of data compression is necessary to get it to the public.

But 96/24 operation doesn't stop just at storage. All equipment in the digital signal chain, including compressors, equalizers, A/D and D/A converters, sample rate converters, and workstations must now be able to process at least 96/24 as well. And since the DVD-Audio, Blu-ray, and HD-DVD formats can also store programs at 192 kHz/24 bit, expect a growing demand for that capability to arise as well.

SURROUND ENCODERS/DECODERS (CODECS)

In the beginning of surround for music (about 1999 or so), most of us thought that we'd have to have a hardware surround encoder hanging around during either the mix or mastering so we could hear exactly what the encoder was doing to the audio. Because there are a lot of parameters that can be tweaked during encoding that can affect the sound (we'll check these out in the next section), we figured that we better take a listen in case something unpleasant happened to the audio that couldn't be fixed later.

The reality of the situation is that encoding took so much time (in the beginning it was at least real time or longer) that it was just impractical to listen to the encode during a mix or even a mastering session. Encoding soon became the domain of the authoring house, and it was usually relegated to the lowest man on the corporate totem pole. As a result, you'll hear many discs with mashed audio (most authoring houses just use the default settings), wrong channel assignments, and a variety of horrors that vex everyone involved in the project as long as the disc is available.

This no longer has to be the case, though, since software encoders that provide speedier results are now available for a reasonable price. Let's bring this process back to where it belongs—the audio people!

So although it's not imperative that an encoder be present during mastering, it does help to hear what the codec (be it some form of Dolby or DTS, SRS, or MLP compressor/decompressor) will do to the final product because codecs can change the sound considerably. There are also quite a few parameters that the producer might like to tweak rather than leaving them for someone else down the production chain.

Data Rate

The biggest change to the audio comes from the data rate selection. In general, the higher the data rate, the closer the encoded signal will be to the source audio (and therefore the better the sound), regardless of the codec that's used. (This only applies to lossy codecs, such as Dolby and DTS.) For Dolby Digital, this means 448 kbps. (Even though 640 kbps is possible, many players won't support it.) For DTS, a data rate of 1,509 kbps is preferred.

The new formats of DTS-HD and Dolby Digital Plus (DD+), TrueHD, and DTS-HD Master Audio blow those data rates away, however. DD+ extends the peak data rate from 640 kbps to 3 Mbps (3,000 kbps), while TrueHD extends it to 18 Mbps (although it is a lossless codec). DTS-HD Master Audio provides a data rate as high as 24.5 Mbps, so it's pretty evident that soon the data rate will be inconsequential to the overall sound quality.

Dialnorm

The purpose of Dialnorm (which stands for *dialog normalization*) is to maintain a consistent dialog level from program to program for the listener. Ever notice how the level changes between the commercials and the program on TV? Or the difference in level from channel to channel (especially cable channels)? This is what Dialnorm was designed to fix, but the idea just never caught on, probably because it wasn't widely understood.

The Dialnorm parameter (know as a *metadata parameter*) is set while encoding and ranges from –31 dB to –1. Believe it or not, the –31 is actually louder than –1 (–31 is the loudest setting), and the default setting is –27! Without getting into the technical reasons why –31 is louder than –1 (it doesn't really matter anyway), if you're encoding music, set it to –31 for the loudest encode, or your client will ask, "Why is the music on my DVD so quiet?"

Data Compression

Data compression is the process of using psychoacoustic principles to reduce the number of bits required to represent the signal. This is needed with surround sound so more data can be squeezed onto a finite storage space, such as a CD or DVD, and also because the bit rate of six channels of 96/24 LPCM, for example, is too large to fit through the small data pipe of a DVD.

Lossy and Lossless Codecs

As stated previously, lossy compression (such as Dolby Digital or DTS) is built around perceptional algorithms that remove signal data that is being masked or covered up by other signal data that is louder. Because this data is thrown away and never retrieved, it's what's known as *lossy*. This is done not only to fit all the data on a disc, but more importantly to fit a lot of data through a small data pipe, especially if it accompanies video (which is a data-rate hog). Think of an inner tube filled up with air. When you let the air out of the tube, it takes up less space. Yet the same amount of rubber remains, and it can fit into a smaller space. This is the same idea behind lossy data compression.

Depending upon the source material, lossy compression can be either completely inaudible or somewhat noticeable. It should be noted that even when it is audible, lossy compression still does a remarkable job of recovering the audio signal, and it still sounds quite good.

Lossless compression (such as MLP) never discards any data and recovers it completely during decoding and playback.

LOSSY CODECS

There are now a number of lossy compression schemes used primarily for DVD encoding from Dolby Digital and DTS (*Digital Theater Systems*).

In general, Dolby Digital (also called *AC-3*, which is actually the file format of the process) compresses the audio data at about an 11:1 ratio to a maximum bit rate of 640 kbps, although 448 kbps is the average data rate used. DTS compresses at about a 3:1 ratio at an average data rate of 1.509 Mbps. Because there is less data compression and therefore less audio data thrown away, many audio professionals prefer the sound of a DTS-encoded product.

Here are all the lossy codecs used on multichannel optical discs (DVD, Blu-ray, HD-DVD) available today:

- **Dolby Digital (.AC3).** Dolby Digital is the standard audio codec for the DVD-Video disc. It's used not so much to save disc space (although it does that nicely), but to send a lot of data when the bandwidth is limited and to leave room for the larger video bandwidth. Dolby Digital (sometimes called AC-3, which is the name of the digital file) takes up to 6 channels (5.1) of 48-kHz/24-bit information.

- **Dolby-EX or DTS-ES.** These are the Dolby and DTS seven-channel, 6.1 audio encoding formats that include a rear center speaker.

- **Dolby Digital Plus.** This is a new audio codec based on Dolby Digital and designed to be backward-compatible with the existing Dolby Digital codec in use today. Dolby Digital Plus is capable of 14 channels (13.1) at a data rate of up to 6 Mbps. Dolby Digital Plus is a standard audio format for HD-DVD video and also an optional format for the Blu-ray disc.

- **DTS Digital Surround.** This is the full name for the audio format standard usually known as just *DTS*. It offers variable compression ratios targeting a wide variety of bit rates and has a base specification that allows for up to 5.1 channels of audio with a 48-kHz sampling rate. DTS is an optional format for the DVD-Video disc and compresses at about a 3:1 ratio at an average data rate of 1.509 Mbps. Because there is less data compression, many prefer the sound of a DTS-encoded product to Dolby Digital, but any differences are greatly dependant upon the program material.

  The company that created the DTS codec, Digital Theater Systems, is co-owned and was co-founded by film director Steven Spielberg, who wasn't satisfied by state of the art in cinema audio when the company was founded. Work on the format started in 1991, but Spielberg debuted the format with his 1993 production of *Jurassic Park*.

  The extensions used for a DTS-encoded file are .cpt, .dts, and .wav. Generally speaking, most newer professional DVD-authoring workstations prefer the .cpt file type, which is somewhat compacted compared to the .dts file. The .cpt file has a marker for the start time of the project. The .wav files are primarily intended for stand-alone audio discs to be used as 5.1 music discs or mixing or mastering check discs.

- **DTS Digital Surround 96/24.** This allows 5.1 channels of 96/24 audio to be delivered on a DVD and has the same bit rate of 1.509 Mbps as DTS Digital Surround. It's also an optional format on both Blu-ray and HD-DVD.

- **DTS Digital-HD** is an extension on the original DTS Digital Surround created for Blu-ray and HD-DVD. It allows for 7.1 channels of 96/24 audio at a bit rate of up to 6.0 Mbps. It's thought to be an option to the lossless DTS Digital-HD Master Audio when space is at a premium.

LOSSLESS CODECS

Lossless audio formats provide compression of about 2 to 1, but no data or fidelity is discarded during compression (which is why it's "lossless"). When uncompressed, the data will be identical to the original.

- **Meridian Lossless Packing.** Meridian Lossless Packing, or *MLP*, is the compression standard used on the DVD-Audio disc in order to store up to six channels of high-resolution 96/24 audio or two channels of 192/24. MLP provides a compression ratio of about 1.85:1 (about 45 percent), and its licensing is administered by Dolby Laboratories.

- **DTS-HD Master Audio.** This is a set of extensions to the DTS Digital Surround audio coding system designed specifically for HD-DVD and Blu-ray. It's capable of up to eight channels of 96/24 or six channels of 192/24 at a bit rate of up to 24.5 Mbps.

Table 10.1 Multichannel Codec Comparison

Codec	Channels	Sample Rate (kHz)	Max Bit Rate (Mbps)	Type
Linear PCM (LPCM)	6	96	27.648	Lossless
Dolby Digital	5.1	48	0.64	Lossy
Dolby Digital EX	6.1	48	0.64	Lossy
Dolby Digital Plus (DD+)	7.1	48	3	Lossy
Dolby True HD	7.1	192	18.64	Lossless
MLP	6	96	9.6	Lossless
DTS	5.1	48	1.509	Lossy
DTS ES	5.1	48	1.509	Lossy
DTS Digital Surround 96/24	5.1	96	1.509	Lossy
DTS-HD Hi-Res Audio	7.1	96	5.76	Lossy
DTS-HD Master Audio	6	192	24.5	Lossless

- **Dolby TrueHD.** This is Dolby's next-generation lossless technology developed for high-definition disc-based media. Dolby TrueHD can support more than eight audio channels of 96/24 audio at up to 18 Mbps bit rate, although HD DVD and Blu-ray disc standards currently limit their maximum number of audio channels to eight.

Surround Software Encoders

To provide an authoring house with an encoded master ready for use on a DVD, HD-DVD, or Blu-ray disc, there are a number of software encoders now available.

MINNETONKA SURCODE

SurCode provides a stand-alone application for encoding Dolby Digital, MLP, and DTS.

NEYRINCK SOUNDCODE

Neyrinck Audio's SoundCode is a plug-in suite of encoders and decoders for the Pro Tools DAW. They market plug-ins for Dolby Digital (including Dolby EX), and DTS Surround, DTS-ES, and DTS-HD.

DTS MASTER AUDIO SUITE

The DTS Master Audio Suite consists of DTS-HD Encoder, SoundCode DTS-HD StreamPlayer, and DTS-HD StreamTools. The Encoder creates all forms of DTS digital audio streams, while the StreamPlayer supports all forms of DTS playback. StreamTools is a tool set designed for encode-stream editing, verification, and bit-stream management.

DOLBY MEDIA PRODUCER

Much like the DTS Master Audio Suite, Dolby Media Producer consists of three very intuitive Mac OS X software applications—the Dolby Media Encoder, the Dolby Media Decoder, and the Dolby Media Tools utility. Each application is stand-alone and very specific in its function, yet supports the full spectrum of Dolby offerings, including Dolby Digital, Dolby Digital Plus, Dolby TrueHD, and MLP Lossless technologies. What's more, Media Producer has the networked facility squarely in mind by providing a complete set of project and file management capabilities.

Media Encoder works with any existing time code or permits embedding new, user-definable code if needed. This means that previously encoded content can be updated using the Punch-In overdub ability that allows you to fix or change time code only in the parts needed, without having to re-encode the whole file. The Media Tools application allows you to repair and update previously encoded files without having to re-encode

them, which really saves a lot of time and deadline anxiety. Among its list of features are file trimming, concatenation (appending files), time-code striping, and the all-important metadata editing.

A New Way of Working

Whereas today's stereo mastering engineers are now used to dealing with the entire mix in terms of adding equalization or compression, surround mastering engineers need more time and expertise to work their magic. For instance, when tweaking the low end (at, say, 60 Hz) the engineer may need to only adjust the LFE channel if that's where the instruments (such as a kick drum) containing that info were assigned. However, it's just as likely that all five main channels, as well as the LFE, will have to be adjusted because the frequency steering by the bass manager to the subwoofer causes that frequency to appear there from multiple sources. This means that whereas the engineer had just one set of stereo adjustments before, multiple adjustments are now needed to accomplish the same thing during for a surround mix.

Surround mastering now also means that the final balance of a mix in terms of level shifts between front and rear speakers and center channel levels are necessary. Out-of-whack LFE levels due to misaligned subwoofers or monitoring without bass management while mixing sometimes require severe adjustment, and, as a result, mastering engineers now require an unprecedented level of control over the final product compared to yesterday's standards.

Other times the mastering engineer might be called upon to create a center channel or LFE channel from the existing program. Or the mastering engineer may be supplied stems and asked to perform a final mix himself. *Stems* are parts of a final mix delivered as separate elements. For instance, a mix of only the rhythm section by itself, the vocals by themselves (complete with effects), and strings or lead elements by themselves would make up three stems that would be mixed together to form the entire mix.

What the Heck Is Authoring?

A DVD, HD-DVD, or Blu-ray disc has a much greater possible level of built-in intelligence than an ordinary CD. *Authoring* is the process of taking advantage of this intelligence by programming not only the interactivity into the disc, but also adding additional material, such as liner notes, music videos, artist and producer bios, and promos for other products.

Because most engineers (and mastering facilities, for that matter) are used to doing the final prep of audio material before either burning a disc themselves or sending it to the replicator, they assume that they will be required to do the same for DVD too. However, this is an area fraught with potential pitfalls that must be approached with caution. Authoring for DVD is a very distant cousin to CD prep, and it is not a trivial matter.

Perhaps the best analogy to DVD (and HD-DVD and Blu-ray) authoring is designing a website. An audio CD is very much like text that you want to send via email. You learn the email program in no time, and soon you're sending mail (burning CDs) worldwide. DVD is more like the World Wide Web. To even put up the most rudimentary site using only text, you've got to program it using HTML. Now if you add pictures, you've got to learn something about graphics or hire a graphic designer to produce something spiffy. If you want to add movies, then you've got to learn about shooting video and video editing and compression, or use an expert.

Nowadays you can buy an inexpensive application that programs HTML for you, but what you get is a very basic, generic site that doesn't compete too well with the big sites that use great graphic designers with intimate coding knowledge to make those advanced web design programs really sing.

As with most professional gear, just buying the authoring workstation does not immediately put you in the authoring game. There is a very high cost of entry for the top-of-the-line systems (you can easily pay well over $50,000 for a workstation with all the necessary peripherals) and a steep learning curve (about six months) before you can get anything out the door in a timely fashion. This is one case where it really is rocket science at the moment, because all of the authoring tools out there have either undocumented or hidden traits that you simply can't learn from a tutorial.

The bottom line is that mastering is *not* authoring and vice versa. Authoring is computer programming that uses the visual, not aural, sense. Unless you have access to design expertise for the graphics, video expertise for video shooting and editing, and programming expertise for the authoring, you're better off leaving the authoring to a facility that specializes in it. Besides, they still need your expertise to supply the best audio possible.

Enter (and Exit) DLT

The current standard, but fading, media used as a DVD production master for delivery to the replicator is DLT (*Digital Linear Tape*), a tape format similar to Exabyte, but with a lot faster transfer rate and greater storage capacity. Since DLT's original use was as a backup medium (with storage of up to 70 GB on a tape), you actually catch a break because the same DLT unit can pull double duty. That is, it can be used for both production master and backup.

DLT is quickly being supplanted by a DVD-R containing a DDP disc image as the replication master.

As we enter this brave new surround mastering universe, it's become obvious that things get pretty complex pretty quickly. As with everything else in recording, only time and experience eventually answer all the questions.

For more information about surround sound production, delivery methods, and calibration, visit the Surround Sound FAQ at www.surroundassociates.com/fqmain.html.

CHAPTER 11

Mastering for Film and Television

Mastering for film and television is the one area of audio where the loudest final audio is not required, or even wanted. In fact, if you deliver audio that is outside the desired specifications (each TV network is a little different), they will kick it back and ask you to do it again.

Although there's not of lot of mastering done specifically for film or television, a mastering engineer may occasionally be asked to supply the final audio for either medium, so it's best to have at least some idea of what those requirements might be. Let's take a look.

Mastering Music for Film

Except on rare occasions, the only thing that gets mastered for film is the musical score or any songs intended for the movie, because the film studio or the production company usually does dialogue and effects. In fact, most of the time the studio does the music as well, but occasionally a recording artist is asked to record the score or songs specifically for a movie, and since the artist feels comfortable continuing his or her normal way of working, the score or songs get mastered.

So the music is mastered as normal and delivered to the dubbing stage, where the dubbing mixer lays it into the movie at the required level. The need for the hottest level doesn't really exist because it will always get adjusted anyway.

On a side note, one of the reasons why the music score for a movie is not normally mastered is that the movie powers-that-be (producer, director, music editor, dubbing mixer) usually ask for the score to be delivered as a 5.1 surround mix with stems. *Stems* are individual submixes of the final mix that allow the dubbing mixer to weave the music around the effects and dialogue so all can be properly heard. Stems are usually delivered as a 5.0 (no LFE channel) mix of the music bed minus the bass, any lead instrument or vocals, and any instruments with a lot of high-frequency information. The bass is then delivered on a separate track, and the lead instrument or vocal and instruments with high-frequency info are each delivered as separate 5.0 mixes (which include all reverbs and ambience). The dubbing mixer then completes the music mix with the rest of the movie.

Mastering for Television

Mastering for television, although not usually requested, is considerably more tricky than mastering for film. Once again, the majority of the time any mastered music audio is delivered to the post-production facility, where it is mixed in against the video. The video editor then determines the correct level against the effects and dialogue, just as in film.

But on the rare occasion when the television audio is coming from the mastering engineer (such as for a concert), the first thing you must do is obtain a technical specification from the engineering department of the network on which it will be shown. This will tell you exactly what they want and how they want it.

Among the types of things that the network specs will contain are all the video requirements (frame size, video levels, video blanking signal, flavor of timecode, color bars, and countdown), as well as the audio requirements. Read and follow these carefully, or you'll end up redoing the project to their liking!

Here's what to watch for:

- The operating level for a reference tone, how long they want the tone, and, if laid back to tape, how far in on the tape it begins. The operating level will usually be –20 dB FS, but sometimes it might be –18 or –16, so check this closely.

- The peak audio levels. (More about this later in the chapter.)

- The acceptable audio quality. (This is actually what they consider unacceptable in terms of distortion and noise.)

- Phasing. (Make sure you listen in mono, because they will.)

- Audio/video synchronization or lip-syncing. A max of usually one frame of lead and a lag of two can be acceptable.

- The desired audio track assignment on the delivery medium.

Of all the above, the peak audio levels are the most important and are usually stated like this: "Programs must have audio levels that regularly peak near but not above –10 dB FS using a peak-reading meter." This means that any peak that goes just a tick beyond –10 will be kicked back for a redo. Keep in mind that the reason that you have to get a spec sheet from each network is that they're all 1 or 2 dB different in this respect, which doesn't seem like a lot until you're spending time redoing it again.

So a mastering job for a movie or a television program is something that happens on a regular basis, but on the rare occasions when you're asked for television delivery, paying close attention to the details will pay off in a lot less hassle.

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Part II

Audio Delivery Formats

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CHAPTER 12

Internet Delivery Formats

Since audio files are very large in their native state, some method of making them smaller must be used in order to send and receive them over the Internet. This method is called *data compression.* At some point in the future when everyone is connected to very high-bandwidth Internet providers, large data files won't be an issue, but for now data compression is the only way for successful online transmission.

Data Compression

Data compression isn't at all like the audio compression that we've talked about previously in the book. Data compression is the process of using psychoacoustic principles to reduce the number of bits required to represent the signal. This is similar to letting the air out of a bicycle tire. It's still a tire, yet you can now fit it into a little box that it couldn't possibly fit into when it was inflated.

Data compression is currently used because the normal LPCM files are so big that they're not easy to transfer or store online. Data compression reduces the amount of physical storage space and memory required to store a sound and therefore reduces the time required to transfer a file.

Data compression can be lossy, meaning the sound quality will be negatively affected by compression, or lossless, meaning there will be no change in sound quality when decoded. Data compression uses a variety of codecs (which stands for *compressor/decompressor*), all with a different sound and a different purpose.

Lossy Codecs

Lossy compression is built around perceptional algorithms that remove signal data that is being masked or covered up by louder signal data. Because this data is thrown away and never retrieved, it's what's known as *lossy*.

Depending upon the source material and the codec parameter settings, lossy compression can be either completely inaudible or somewhat noticeable and objectionable. It should be noted that even when it is audible, lossy compression still does a remarkable job of recovering the audio signal and can still sound quite good.

- **MP3 (officially known as MPEG-1 Audio Layer 3).** The MP3 file (.mp3) is a common compressed WAV file. MPEG-1 files are about one-twelfth the size of WAV files. This is why MP3 players can accommodate thousands of songs on a tiny chunk of storage space.

- **AAC.** This stands for *Advanced Audio Coding*; it was developed by the MPEG group that includes Dolby, Fraunhofer (FhG), AT&T, Sony, and Nokia—companies that have also been involved in the development of audio codecs such as MP3 and AC3 (see Dolby Digital). For a number of years, many cell phones from the big manufacturers, such as Nokia, Motorola, and Sony Ericsson, have supported AAC playback. Sony has also added support for playing back AAC files on its PSP player as well.

  AAC can have better audio quality than MP3 at equivalent or slightly lower bit rates. Here is a list of just some of the advantages AAC has over MP3 (even when the MP3 is encoded with the latest LAME encoder):

  - Sample frequencies from 8 Hz to 96 kHz. (MP3 is 6 Hz to 48 kHz.)

  - Up to 48 channels.

  - Higher coding efficiency, which means better quality at a lower bit rate.

  - Much better handling of frequencies above 16 kHz.

  - Better handling of transients.

  AAC is wrapped in the MPEG-4 container (.mp4, .m4a, and so on) and is rapidly gaining support. The Apple iPod is fully compatible with AAC in MPEG-4.

- **MPEG-4.** First, it's important to understand that MPEG-4 is a new standard and has nothing to do with MP3. The MPEG-4 technology works by splitting content into its individual elements. A small movie, for instance, can be seen as audio, video, titles, and subtitles—four different elements that together form the complete movie. If you want the best quality using the least amount of disc space, you need to analyze each of these elements and choose the appropriate compression format for each. For example, if in the movie someone is only making a phone call, you could use an audio compression format that needs less quality than when you see an orchestra playing in an opera house in the same movie. If the person is making a phone call and he only moves his lips, you need less movie quality than when you're showing an entire moving orchestra playing a powerful song.

  MPEG-4 has several different extensions:

  - **.mp4** The official extension for both audio and video files.
  - **.m4a** Introduced by Apple for Apple Lossless Audio Coding files, m4a can safely be renamed to .mp4.
  - **.m4p** Digital Rights Management (DRM)–protected files sold on iTunes.
  - **.m4e** Renamed .sdp files used by Envivio for streaming.
  - **.m4v**, **.mp4v**, **.cmp**, **.divx**, **.xvid** Video-only, raw MPEG-4 video streams.
  - **.3gp, .3g2** Used by mobile phones. Also stores content not defined in .mp4.

- **Windows Media Audio.** Windows Media Audio (.wma) is a proprietary compressed audio file format developed by Microsoft. It was initially developed as a competitor to the MP3 format, but with the introduction of Apple's iTunes Music Store, it has positioned itself as a competitor to the Advanced Audio Coding format used by Apple. A large number of consumer devices, ranging from portable handheld music players to portable CD players and set-top DVD players, support the playback of WMA files.

  The most current version of the format (WMA9) includes specific codecs for lossless, multichannel surround sound, and voice encoding in addition to the main lossy codec. Both constant and variable bit rate encoding are supported.

A WMA file is almost always encapsulated in an Advanced Systems Format (ASF) file. The resulting file may have the file extension .wma or .asf, with the .wma extension being used only if the file is strictly audio. The ASF file format specifies how metadata about the file is to be encoded, which is similar to the ID3 tags used by MP3 files. ASF is also patented in the United States.

- **Ogg Vorbis.** Ogg Vorbis (.ogg) is another compressed source code similar to MP3, but like WMA it is more efficient at data compression, so the files are smaller. Ogg Vorbis is also open source (free to all, unlicensed, no strings attached). While most MP3 encoders compress data at a constant bit rate, Ogg uses a variable bit rate. This means that if you are copying chunks of silence into MP3 format using a constant bit rate, the compression bit rate stays the same as if you were compressing the sound of an entire orchestra. But if you are copying chunks of silence into Ogg, the rate varies with the need, and your data rate will drop to nothing.

- **µ-law.** The µ-law (pronounced *mu-law*) file format is an international standard for compressing voice-quality audio. It has a compression ratio of 2:1. Because it's optimized for speech, in the United States it is a standard compression technique for telephone systems. (In Europe its cousin A-law is used.) On the Internet it uses the .au file formats, alternately know as *Sun audio* formats. The A-law algorithm provides a slightly larger dynamic range than the µ-law at the cost of worse proportional distortion for small signals. By convention, A-law is used for an international connection if at least one country uses it.

Lossless Codecs

Unlike lossy codecs, lossless codecs don't throw away data to make the file smaller, and, as a result, they sound a lot better. They're generally larger than lossy codecs, though, and take longer to encode.

- **Apple Lossless.** Apple Lossless (also known as *Apple Lossless Encoder*, *ALE*, or *Apple Lossless Audio Codec*, *ALAC*) is an audio codec developed by Apple Computer for lossless encoding of digital music. Apple Lossless data is stored within an MP4 container with the filename extension .m4a. ALAC-compressed files are about 60 percent of the size of the originals, similar to other lossless formats. Compared to most other formats, Apple Lossless is not as difficult to decode, making it practical for a limited-power device, such as an iPod. The Apple Lossless Encoder was introduced as a component of both QuickTime and iTunes.

- **FLAC.** FLAC (*Free Lossless Audio Codec*) supports linear PCM samples with resolutions between 4 and 32 bits, sample rates from 1 Hz to 1,048,570 Hz in 1-Hz increments, and from one to eight channels per stream separately or, if required, multiplexed together in a suitable file container.

  With FLAC, you do not specify a bit rate as you do with some lossy codecs. The resulting bit rate is roughly proportional to the amount of information in the original signal, and the result can be from around 100 percent of the input rate (if you're encoding a spectrally dense sound, such as noise) down to almost 0 when you are encoding silence. FLAC is stored with a .flac extension.

Streaming Audio

Streaming audio avoids many of the problems of large audio files. Instead of having to wait for the entire file to download, you can listen to the sound as the data arrives at your computer. It's also a very secure method of transmission for the artist and the record label because the file is never downloaded.

Streaming audio players store several seconds of data in a buffer before beginning playback. The buffer absorbs the bursts of data as they are delivered by the Internet and releases them at a constant rate for smooth playback.

Many digital audio formats can be streamed by wrapping them in a streaming format, such as Microsoft's ASF (*Active Streaming Format*), which can be used to stream MS Audio, MP3, and other formats.

Table 12.1 shows several streaming audio formats.

Table 12.1 Streaming Audio Formats

Type	Primary Format	Developer
Windows Media Technologies	Windows Media Audio/Active Streaming Format (ASF)	Microsoft
Icecast (open source)	MP3	The Icecast Team
QuickTime	QuickTime	Apple Computer
RealSystem	RealAudio	RealNetworks
SHOUTcast	MP3	Nullsoft

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CHAPTER 13

Optical Discs: CDs

When CDs were first introduced, neither the disc nor the player had the intelligence that the later DVD, Blu-ray, and HD-DVD formats have. So, in order to provide a disc capable of many uses, a number of CD formats known as *Books* were created. Some of these never caught on, and some are only occasionally used today, but it helps to have the information in one spot if you ever need it. So here it is—everything you ever wanted to know about CDs, plus a reference list to find out even more at the end.

The Books

When it comes to the technical talk about CDs, sooner or later the matter of Books comes up. The Books are simply sets of technical specifications that CDs must follow to be compatible with each other and therefore to be able to play on any player. Because quite a large number of books exist, it's easy to get overwhelmed and confused, but they're really quite simple once you get rid of the technical jargon.

RED BOOK

Red Book is the prerecorded CD audio standard that you find in music stores today. Because of this standard, any audio CD will play in any audio compact disc player, and this has been a major factor in the growth of the CD industry. Specified are the sample rate (44.1 kHz), bit depth (16), type of error detection and correction, and how the data is stored on the disc, among other things.

Also defined is a way to add graphics information to the CD for a CD+G (*CD plus Graphics*) disc, which was weakly tried by the major record labels in the mid '80s and is not generally available today. Approximately 16 MB of graphics data can be stored on a disc. Each Red Book disc can have up to 99 audio tracks and can be 74:33 minutes in length (although it's possible to reach 80 minutes under special circumstances).

ORANGE BOOK

The Orange Book defines the standard for writable or recordable media, such as CD-Rs, rewritable CD-Rs (CD-RW), and Magneto Optical discs (another disc format that never caught on). It defines where the data can be written and, in the case of the MO, how it is erased and rewritten.

BLUE BOOK

This is a hybrid disc that is part Red Book and part Yellow Book. A Blue Book CD is also sometimes referred to as *CD Plus* or *CD Extra*.

An offshoot of a Blue Book/CD Extra disc is an enhanced CD. The difference is the order in which the files are written, which is data first (the Yellow Book info), then audio in the CD Extra.

GREEN BOOK

A precursor to DVD in terms of flexibility, the Compact Disc Interactive (CD-I) standard was released by Philips in 1987 and allows for full-motion video on a standard 5" disc. Now defunct, it requires a dedicated CD-I player and is not compatible with a standard audio CD player.

YELLOW BOOK

This is the CD-ROM standard for computer data. It also adds two additional track types that differ from the Red Book audio disc—Mode 1, which is usually computer data, and Mode 2, which is usually compressed audio data or video/picture data.

WHITE BOOK

Sometimes known as *Karaoke CD*, White Book CDs are used in applications in which the combination of limited full-motion video and audio is needed. These were originally called *Video CDs*, but they were soon renamed due to the more widespread use in karaoke applications. White Book CDs utilize MPEG 1 and 2 compression schemes in order to compress audio and video down to a usable size. The format was originally written by Philips in conjunction with the Japanese Victor Company (JVC) and is also supported by Sony and Matsushita.

PHOTO CD

Developed by Eastman Kodak and Philips, Photo CD is a way of cataloging photographs on a CD. The photos can be read in a number of ways—from a dedicated photo CD player, from CD-I players (now obsolete), from CD-ROM on a computer with a Photo CD driver set, and from 3DO players.

SCARLET BOOK

Basically an extension of the Red Book, Scarlet Book is the official specification of the Super Audio CD (see the “The Super Audio CD (SA-CD)” section in Chapter 14) and was the last Book specification created.

CHAPTER 14

Optical Discs: Multichannel Delivery

Now that the DVD-Video disc has been around for a while and the HD-DVD and Blu-ray formats are becoming commonplace on the shelves of the local electronics superstore, there are a few audio specialty formats that, while quickly fading from view, are still available. The DVD-Audio disc, Super Audio CD (SACD), and, prior to that, the DTS Music Disc were once touted as the saviors for music. While they were somewhat accepted by audiophiles, the consumer public met them with a collective shrug of indifference. And although the audio quality with these discs can be superior to other formats, the primary attraction has been the fact that they provide multichannel delivery (surround sound). This chapter presents a quick but thorough overview of each format.

DVD Basics

Most of the formats that I'll be discussing are in some way based upon the DVD (sometimes mistakenly but appropriately named *Digital Versatile Disc*) concept, so some DVD basics are in order. A DVD distinguishes itself from a CD in two ways—storage capacity and file format.

STORAGE CAPACITY

While the storage capacity of a typical CD is 700 MB (with 800 MB available but rarely used), the capacity of a DVD can actually be one of four levels, all far exceeding the CD. This is accomplished by having more and smaller pits on the substrate than those on a CD. Add to this the fact that DVD can have two layers and be double-sided, and the power of DVD

becomes readily apparent (see Table 14.1). Because a laser with a smaller wavelength is required, a CD player cannot read a DVD. A DVD player can read a CD, though.

Table 14.1 DVD Types and Capacities

DVD Name	Number of Sides	Number of Layers	Capacity
DVD-5	1	1	4.7 G bytes
DVD-9	1	2	8.5 G bytes
DVD-10	2	1	9.4 G bytes
DVD-18	2	2	17 G bytes

Note: There's some unfortunate confusion as to the actual capacity of a DVD because it's measured differently than the computer norm. For example, a DVD-5 has 4.7 billion bytes (G bytes), not 4.7 gigabytes (GB). The problem is that DVD is based on multiples of 1,000, whereas the computer world measures bytes in multiples of 1,024. Therefore, a DVD-5 actually has a capacity of 4.38 GB.

FILE FORMAT

Today's CD can be thought of as essentially a "bit bucket" in that there is no intelligence built into the different file formats required for audio CD, CD-ROM, CD-R, and so on. DVD differs in that the various types use basically the same DVD-ROM-like format with a bit of intelligence built into the specification.

DVD uses a file format known as *Universal Disc Format*, or *UDF*, which was designed specifically for use with optical media and avoids the problems and confusion that CD-ROMs had because of the many different competing file formats used. In fact, UDF permits the use of a DVD by DOS, OS/2, Macintosh, Windows, and UNIX operating systems, as well as dedicated players. What's interesting is that a dedicated DVD player will access only the information that it requires, and all other files will remain invisible. It also means that the file system for use with computers is already built into the format, which widens the potential market without you having to jump through programming hoops.

The DVD-Video Disc

DVD-Video burst onto the scene in 1998 primarily as a high-quality movie delivery system, but the audio portion of the format is still quite an improvement over the Red Book CD standard. And because there are automatic provisions for multichannel audio and a built-in (but limited) 96/24 option, DVD-V is occasionally used as a delivery format for audio.

DVD-V AUDIO SPECS

The audio portion of a DVD-V can have up to eight bit streams (audio tracks). These can be one to eight channels of common linear PCM (LPCM), one to six channels (5.1) of Dolby Digital, or one to eight channels (5.1 or 7.1) of MPEG-2 audio (see Table 14.2). Also, there are provisions for optional DTS or SDDS encoding.

Table 14.2 Audio Portion of a DVD-Video

Audio Coding	Sample Rate (kHz)	Word Length	Number of Channels	Max Bit Rate
LPCM	48 48 48 96 96 96	16 20 24 16 20 24	8 6 4 4 3 2	6.144 Mbps
Dolby Digital	48	24	6	448 kbps
MPEG-2	48	16	8	912 kbps
DTS (optional)	48	20	6	1.4 Mbps

The LPCM bit stream, which is the same uncompressed format as the typical Red Book CD (which is standardized at 44.1 kHz and 16 bits), can use either a 48- or 96-kHz sample rate with a bit depth of either 16, 20, or 24 bits. Now on the surface this seems great and makes you wonder why another format for multichannel audio is even considered, but then you realize that the bit rate for the audio data is capped at 6.144 million bits per second (Mbps).

The bit rate (the sample rate times the number of bits times the number of channels) is equivalent to the size of the pipe that the audio data has to flow through, and in this case the pipe isn't big enough to fit six channels of 96/24 audio. In fact, all you can squeeze through is two channels of 96/24. If you want multichannel, you're back at 48k, but at least the bit depth is raised to 20 bits for six channels (refer to Table 14.2). So now you have to use some sort of data compression scheme to fit all of the channels down the pipe at a higher audio quality.

The standard compression scheme for DVD-V is Dolby Digital (sometimes called *AC-3*, which is actually the name of the file format after it has been compressed), which compresses six channels (5.1) of up to 24-bit audio to fit through the DVD-V audio pipe, but is limited to only a 48-kHz sampling rate. Plus it's a lossy compression algorithm with a maximum bit rate of 640 kbps (although 448 kbps is mostly used), which

means that some data is thrown away in the encoding process (although the goal is to only throw away the data that you won't miss). MPEG-2 Audio, which can be configured either six-channel (5.1) or eight-channel (7.1) at 48/16, is also an optional compression scheme, but it is hardly ever used (especially in the U.S.) due to a lack of decoders in the marketplace. Even though MPEG-2 does have a higher bit rate at 912 kbps, the algorithm has its share of inherent coding problems, which effectively negates its lower data compression ratio.

While Dolby Digital is the default encoding process of the DVD-Video disc and at least one track must use it (a short menu will do), DTS can also prove to be an interesting choice because it can encode up to six channels with less data compression than either Dolby Digital or MPEG. (See "The DTS Music Disc" section later in this chapter for more details.)

DVD-V VIDEO SPECS

DVD-Video uses the MPEG2 codec to support a 720×480i video resolution. MPEG1 is also allowed, but rarely used.

DVD-VIDEO ADVANTAGES

- **Installed base of players.** DVD-V audio can currently play on all DVD players in the marketplace and all computer DVD-ROM drives as well, provided that the PC has the appropriate decoding hardware/software.

- **Compatibility with the greatest number of players.** Unlike DVD-A, which requires a player with specific playback capability, DVD-V audio is universally compatible with existing and future players, such as HD-DVD.

DVD-VIDEO DISADVANTAGES

- **96/24 LPCM available on only two channels.** The highest quality multichannel LPCM audio available is 48 kHz at 20 bits for six channels. Using a data compression scheme, such as Dolby Digital, gives you six channels of 48/24.

- **Some players can't handle 96/24 LPCM.** Even if 96/24 LPCM is used, some players automatically decimate to 48 kHz and truncate to 16 bits (although they don't tell you they're doing it), thereby negating some of the benefits of the enhancement.

The DTS Music Disc

There's some confusion in the marketplace as to exactly what DTS (*Digital Theater Systems*) is. Is it a company? Is it a technology? Is it for movies? Is it for music? The answer is really yes to all of the above.

DTS the company was started in 1994, primarily with the intention of bringing higher quality audio in surround sound to motion pictures than what was available at the time. This was done by way of the DTS data compression process, which is a lossy data compression that reduces the data less and with a different method than its competitor, Dolby Digital. The DTS compression scheme supposedly sounds better as a result. To prevent confusion, the codec is now called DTS Digital Surround.

This data-compressed film audio was then burned to a CD, synced to the film, and translated back into analog 5.1 audio in the theater via a hardware decoder. Since putting audio on a disc was already being done by DTS for film sound, the next logical step was to make a CD strictly for commercial distribution of surround-sound music. Hence the DTS music disc was born.

The DTS music disc is actually the only multichannel delivery system of the six discussed that isn't based in some way on the DVD spec. In fact, the DTS-compressed bit stream is encoded onto what amounts to a CD-ROM. This can then be played back on any CD player, laser disc player, or DVD player that has a digital output and passes the digital bit stream to a DTS decoder that separates the channels back out to 5.1.

To promote their technology, DTS started their own record company called DTS Entertainment to license previously released and new recordings remixed in surround, which would help promote the format.

The problem with the DTS music disc, however, was that people bought the discs thinking that they were buying a normal CD, and when they tried to play them back, they got a hail of white noise out of their speakers. This was because the digital stream on the disc (the DTS encode) needed to be played out via the digital output of the player and decoded by a receiver. When you played it out the analog outputs (as most consumers are used to doing), all you heard was this horrible noise. As a result, most retailers pulled the disc from their shelves because they weren't able to educate potential buyers as to what the disc actually was.

Although the discs are still released today, they are relegated to the audiophile outlets where the buyer has a better understanding of the technology. It's also a nice format to use as a check disc for a surround mix.

DTS MUSIC DISC AUDIO SPECS

The DTS music disc provides up to 74 minutes of 5.1 audio at a sample rate of 44.1 kHz or stereo at 88.2 kHz. It will only accept a 20-bit source, but at the relatively high bit rate of 1.4 Mbps. As stated before, the big attraction to DTS is the fact that the compression algorithm uses a gentle 3:1 ratio, which many claim sounds better as a result.

Later disc releases also use the DTS Digital Surround 96/24 codec to get the full 24 bits on the disc.

DTS MUSIC DISC VIDEO SPECS

The DTS music disc does not support video.

DTS MUSIC DISC ADVANTAGES

- **Sonic superiority.** Thanks to its low compression ratio and high bit rate, many audio professionals (but not all) feel that the DTS encoding system sounds the best of the current lossy compression systems.

- **A sizable catalog.** A wide library (several hundred discs in all musical genres already released) of DTS music discs can be found both online and at specialty audio retailers.

DTS MUSIC DISC DISADVANTAGES

- **Requires a decoder to operate.** Without a DTS decoder, the only output you get from your player or receiver is white noise. However, almost all receivers, even the most inexpensive ones, now come with a DTS decoder built in.

- **Distribution limited due to non-compatible discs.** Because of possible consumer confusion with Red Book CDs (the customer puts it in his CD player, only to get a white noise output), many of the biggest music retailers have refused to carry the DTS music disc to this point.

- **No value-added information.** Because the DTS music disc uses the limited storage capacity of a CD, there's no room (or provision) for additional text, graphics, or video material.

The DVD-Audio Disc

Introduced in mid-2000 after several years of preparation, the DVD-Audio disc (DVD-A) provides significantly higher audio quality than its video cousin. Just having the ability to do so doesn't necessarily mean that the highest fidelity audio will automatically happen, though, because for better or for worse, the final decision as to the sonic quality is largely in the hands of the content producer.

DVD-A AUDIO SPECS

DVD-A differs from the audio portion of DVD-V in that the data pipe is a much larger 9.6 Mbps compared to DVD-V's 6.144 Mbps. Even with the wider audio pipe, six channels of 96/24 LPCM audio still exceeds the allotted bandwidth. (Multiply 96k by 24 bits by six channels to get the resultant 13.824-Mbps bandwidth.) Therefore, there needs to be some type of data compression to not only fit the required amount of data through the pipe, but to increase the playing time as well.

For this requirement, Meridian Lossless Packing (MLP) was selected as the standard data compression for DVD-A. MLP, which provides about a 1.85:1 compression ratio, is lossless, meaning that no data is thrown away during the compression process. Dolby Digital is listed as a lossy compression option. Also possible is the use of other coding technologies besides LPCM, such as DTS.

SCALABILITY

One of the more interesting, but possibly confusing, traits about DVD-A is what's known as *scalability*, which simply means "lots of options." Audio-wise those options are extensive. The program producer is able to choose the number of channels (one to six), the bit depth (16, 20, 24), and the sample rate (44.1, 48, 88.2, 96, 176.4, or 192 kHz). See Table 14.3.

Table 14.3 DVD-A Audio Scalability

Audio Coding	Sample Rate (kHz)	Word Length	Number of Channels
LPCM	192 176.4 96 88.2 48 44.1	16, 20, 24 16, 20, 24 16, 20, 24 16, 20, 24 16, 20, 24 16, 20, 24	2 2 1 to 6 1 to 6 1 to 6 1 to 6
MLP	96	16, 20, 24	1 to 6
Dolby Digital	16, 20, 24	1 to 6	
DTS	1/2	16, 20, 24	1 to 6

In theory, the producer can also mix and match different sample rates with different bit depths on different channel families. For example, the front three channels (family 1) can be set to 96/24, while the rear (family 2) and sub channels are set to 48/16. Although this might be important for more efficient bit budgeting if additional space for videos or stereo mixes is required, the feature isn't used much in everyday practice.

PLAYBACK TIME

Even with DVD-A's increased storage capacity, there's still not enough room to contain 74 minutes of discrete multichannel linear PCM (LPCM) program at the high sample rates and bit depths. So the option exists to compress the audio data in several ways.

As stated before, for the high sample rates and bit depths (88.2, 96, 176.4, or 192 kHz/24 bit), Meridian Lossless Packing, or MLP, is provided. This method is attractive in that it almost doubles the playing time with no loss in data and therefore audio quality. For the lower sample rates and bit depth (48k/20 bit), Dolby Digital (AC-3) is also provided as an option.

COPY PROTECTION AND WATERMARKING

Of primary concern to all the committees and groups working on DVD-A was the inclusion of strong anti-piracy measures and copyright identification. In fact, the encryption and watermarking issues actually took the longest to resolve and held up the release of the format longer than any other technical aspect. This proved to be an almost fatal blow, since by the time DVD-A was finally released *en masse*, DVD-Video had taken over the consumer consciousness, and the MP3 boom was just around the bend. In the end, watermarking was never incorporated, and copy protection was little used.

VALUE-ADDED CONTENT

One of the attractive features of DVD-A is the ability to add additional content, such as liner notes, music videos, and additional video features. This could prove another immediate advantage because consumers have always complained about the lack of information found on CDs. Couple this with additional artist commentary, discographies promoting back catalog titles, bios, links to websites (and therefore aftermarket sales), and even a place to finally put those videos that MTV never played, and the value-added material brings the format to life.

Each track (song) has the ability to display up to 99 still images that can run like a slideshow in an automatic or manual mode. This can be either a great way to display artist or song information or a lame attempt to add some info that no one wants to see, depending on how it's implemented. Videos can also be added in the video portion of DVD-A (there is always a video zone) provided that there's sufficient room left on the disc.

DVD-A VIDEO SPECS

There is a video zone available on a DVD-A that uses the exact same video spec as a DVD-V, which is MPEG2 at a resolution of 720×480i.

DVD-AUDIO ADVANTAGES

- **Scalability.** The program producer, mastering engineer, or author is able to choose the number of channels, the bit depth, the sample rate, and the encoding method.
- **Value-added material.** Liner notes, album-cover artwork, music videos, and artist commentary can all be included.

DVD-AUDIO DISADVANTAGES

- **A limited market.** It's a largely failed format with a very small audience.
- **Lack of moving pictures during the song.** Many in the production community believe this to be a more important feature than high-quality audio, even though up to 99 still pictures per song may be used.
- **DVD-A discs require a compatible player.** Since DVD-A was specified well after DVD-V hit the marketplace, DVD-A discs cannot play on early-generation players. Many, but not all, new players have the ability to play any DVD format (called a "combi" player). This limitation can be sidestepped by authoring the video zone of the disc with identical, albeit lower-quality (48/24 Dolby Digital–encoded), material, but it's up to the content owner to provide this added authoring.

The Super Audio CD (SA-CD)

Although the promise of vastly improved sonic performance as well as backward and forward compatibility make the Super Audio CD (SA-CD) an intriguing prospect in the multichannel delivery wars, it became yet another failed format with about the same market penetration as the DVD-A. Even with the massive corporate muscles of Sony and Philips behind this format, SA-CD was never seriously considered as a replacement for the CD by the marketplace, and the product that was released was scaled back in terms of features from what was originally announced.

The SA-CD can be made as a dual-layer disc (basically a DVD-9, known as a *hybrid*) with one layer dedicated to normal Red Book CD–type audio and the second to a high-density layer for a six-channel surround mix or a high-resolution two-channel stereo mix. What made this interesting to the record labels is the ability to be both backward and forward compatible, meaning that consumers can play an SA-CD on their current CD player as if it were a normal CD, and also play a current CD on an SA-CD player as well. SA-CD discs are not playable in existing DVD-ROM drives, however.

SA-CDs are actually an extension of the Red Book CD known as the *Scarlet Book* (although sometimes called the *Crimson* or *Burgundy Book*). Many are shipped in a distinctive package called a *Super Jewel Box* that is larger than a normal CD jewel box and has rounded edges.

SA-CD AUDIO SPECS

SA-CD touts an improvement in sonic quality due to a recording process known as *Direct Stream Digital* (*DSD*). DSD uses essentially the same delta-sigma oversampling method used in most modern high-quality analog-to-digital conversion systems, where a single bit measures whether a waveform is rising or falling rather than measuring an analog waveform at discrete points in time. In current systems, this one bit is then decimated into LPCM, causing a varying amount (depending upon the system) of unwanted audio side effects (such as quantization errors and ringing from the necessary brick-wall filter). DSD simplifies the recording chain by recording the one bit directly, thereby reducing the unwanted side effects.

Indeed, on paper SA-CD with DSD looks impressive. A sampling rate of 2.8224 MHz (which is 64 times 44.1k, in case you're wondering) yields a frequency response from DC to 100 kHz with a dynamic range of 120 dB. Most of the quantization error is moved out of the audio bandwidth, and the brick-wall filter, which haunts current LPCM systems, is removed. To enable a full 74 minutes of multichannel recording, Philips has also developed a lossless coding method called *Direct Stream Transfer* that provides a 50-percent data reduction. DSD is a closed system that has shown little room for improvement in that both the frequency response and dynamic range have not improved much beyond the initial specs, and there are few interfaces, DSP chips, and supporting software compared to their LPCM counterparts.

SA-CD VIDEO SPECS

The SA-CD format does not support video.

For more information about SA-CD, go to SA-CD.net.

SA-CD ADVANTAGES

- **Sonic performance.** Wide bandwidth goes from DC to 100 kHz with a 120-dB dynamic range. There are no adverse filter artifacts thanks to elimination of the brick-wall filter. There have been widespread positive reviews regarding audio quality.
- **Plays on current CD players.** With both backward and forward compatibility, consumers don't feel forced to buy expensive new hardware or give up their current libraries.

SA-CD DISADVANTAGES

- **A limited market.** As with DVD-A, the average consumer was not willing to buy another piece of expensive hardware and was generally confused with yet another format choice.
- **Is the sonic performance really better?** While DSD seems every bit the equal to the current state of LPCM, advances in converter technology could eventually move LPCM beyond the seemingly closed format of SA-CD.

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CHAPTER 15

Optical Discs: The High-Resolution Discs

So how do the HD-DVD and Blu-ray (also known as *BD*) formats differ from the common DVD? All the high-resolution, high-definition features are the result of one thing—the pits that contain the encoded data are a lot smaller than on a DVD, and the laser (a blue-violet one as opposed to a red one) is a shorter wavelength that makes it possible to focus with greater precision. This allows data to be packed more tightly and stored in less space, so it's possible to have more storage capacity on the disc even though it's the same physical size as a CD or a DVD. This also provides the increased bandwidth and data rate needed for high-definition video and the resulting high-sample-rate multichannel audio.

Although Blu-ray and HD-DVD are similar in many aspects, there are some important differences between them. Blu-ray holds up to 25 GB per layer (50 GB on a dual-layer disc), whereas HD-DVD holds up to 15 GB and 30 GB on a dual-layer disc. Blu-ray has also adopted a higher transfer rate for video and audio (54 Mbps versus 36.55 Mbps), but HD-DVD has standard features such as interactivity, network accessibility, and a secondary codec that Blu-ray is only now about to incorporate as an option.

HD-DVD

The High-Definition DVD, or HD-DVD, is a high-capacity optical disc specifically designed to be the successor to the standard DVD. It was jointly developed by NEC and Toshiba, and the standard was approved by the DVD Forum, the same association of companies that set the standards for the DVD.

HD-DVD OVERVIEW

HD-DVD holds up to 15 GB on a single-layer disc and 30 GB on a dual-layer disc, and it has a data rate of 36.55 Mbps (compared to DVD's 9.8 Mbps). This means that both high-definition audio and video are possible at the same time. Like the DVD, HD-DVD utilizes the Universal Disc Format (UDF) file system. Because HD-DVD was built around the standard DVD specs, the players are relatively simple and inexpensive to build (and retail cheaper as a result), and the discs are easy to replicate. Also, HD-DVD players are backward compatible, so standard DVDs can play in them. HD-DVD can provide a playback time of up to four hours of high-def program on a 15-GB disc and eight hours on a 30-GB disc.

HD-DVD burners and discs are said to be available, but not in large quantities. As a result, widespread use by computer manufacturers has not taken place. Pre-recorded titles are available in abundance, although not as many as Blu-ray at this time (late 2007).

HD-DVD AUDIO SPECS

HD-DVD supports almost every audio codec format, but because of the high data rate, it's now possible to store up to eight channels of 96/24 LPCM audio without data compression. Almost all codecs, lossy and lossless, can be used, especially if video is present on the disc.

Table 15.1 presents a list of the mandatory and optional audio formats. *Mandatory* means that all HD-DVD players are required to decode the format. A secondary audio track, if present, can use any of the mandatory formats or one of the optional codecs.

HD-DVD VIDEO SPECS

HD-DVD supports the 720×480i video resolution of the DVD-Video standard, and uses the MPEG2 codec for this function. But it really shines with the new, more efficient SMPTE VC-1 (which is based on Microsoft's Windows Media 9 codec) and MPEG4-AVC video high-def codecs, with resolutions of 720P, 1080i, and 1080p.

Table 15.1 HD-DVD List of Codecs

Codec	Channels	Sample Rate	State	Type
Linear PCM (LPCM)	Up to 8	4	Mandatory	Lossless
Linear PCM	2	8	Mandatory	Lossless
Dolby Digital	5.1	2	Mandatory	Lossy
Dolby Digital EX	6.1	2	Mandatory	Lossy
Dolby Digital Plus (DD+)	7.1	2	Mandatory	Lossy
Dolby TrueHD	7.1	4	Mandatory	Lossless
Dolby TrueHD	2	8	Mandatory	Lossless
MLP	6	4	Optional	Lossless
MLP	2	8	Optional	Lossless
DTS Digital Surround	5.1	2	Mandatory	Lossy
DTS Digital Surround ES	6.1	2	Optional	Lossy
DTS Digital Surround 96/24	5.1	4	Optional	Lossy
DTS-HD High-Res Audio	7.1	4	Optional	Lossy
DTS-HD Master Audio	Up to 8	4	Optional	Lossless
DTS-HD Master Audio	Up to 6	8	Optional	Lossless

HD-DVD ADDITIONAL FEATURES

HD-DVD offers many additional features that tower over DVD-Video. All of these features are mandatory for HD-DVD players. Some of these are:

- **The ability to network players.** Players can be connected to a network for firmware updates and web-enabled content.

- **A secondary video decoder.** A secondary video decoder allows for two simultaneous video streams that make picture-in-picture possible. This can also be used for things such as directors' commentaries that can be watched during the movie itself, or it can be used to show the difference between HD and SD (Standard Definition) side by side.

- **Persistent storage.** This allows titles to save information in the player (such as bookmarks) and user preferences (such as language choice).

- **Interactivity.** This provides you with the ability to interact with a title via picture-in-picture, games, menus, unlocking disc content, or accessing web content. HD-DVDs use what's called *HDi Interactive Format* to author interactive content, which is based on familiar web programming languages such as HTML, XML, and CSS. This makes for an easy learning curve for a web developer versed in those scripting environments.

HD-DVD ADVANTAGES

- **High-resolution audio and video.** The high data rate and storage capacity make a wide range of resolution options possible.
- **Inexpensive to make.** Because the technology is based around the widespread DVD format, both hardware and software are relatively inexpensive.
- **Additional features.** Features such as persistent storage in players, secondary codecs, network accessibility, and interactivity are powerful extra features not currently mandatory on any other optical disc format.
- **Easier to program.** Because it's built on a previously proven technology and operating system, both main and interactive content are easier to program than Blu-ray.

HD-DVD DISADVANTAGES

- **Less industry support.** Because stand-alone burners have been slow to market, computer companies (except the Toshiba, one of its creators) have been slow to pick up on HD-DVD. Pre-recorded titles also lag behind Blu-ray.
- **Lower bandwidth than Blu-ray.** Because the bandwidth is lower, it is argued that the resultant quality of the high-def output is not as good as Blu-ray. However, since each format uses the same codecs, this is highly debatable.
- **Lower storage capacity than Blu-ray.** HD-DVD has a capacity of 15 GB versus 25 GB for Blu-ray. (The dual-layer figure is doubled for both.) This may not be much of a problem for pre-recorded titles, though.

Blu-ray

Blu-ray (BD) is the name of the optical disc format initially developed by Sony and Philips (inventors of the compact disc, cassette, and laserdisc) as a next-generation data- and video-storage format alternative to DVD.

The format was developed to enable recording, rewriting, and playback of high-definition audio and video, as well as storage of large amounts of data. It offers more than five times the storage capacity of traditional

DVDs and can hold up to 25 GB on a single-layer disc and 50 GB on a dual-layer disc.

Blu-ray is currently supported by more than 180 of the world's leading consumer electronics, personal computer, recording media, video game, and music companies, including Sony, Apple, Dell, Hitachi, HP, JVC, LG, Mitsubishi, Panasonic, Pioneer, Philips, Samsung, Sharp, Sony, TDK, and Thomson. The format also has broad support from the major movie studios as a successor to today's DVD format. In fact, seven of the eight major movie studios (Disney, Fox, Warner, Sony, Lionsgate, and MGM) have released movies in the Blu-ray format, and five of them (Disney, Fox, Sony, Lionsgate, and MGM) are releasing their movies exclusively in the Blu-ray format.

The name *Blu-ray* is derived from the underlying technology, which utilizes a blue-violet laser to read and write data. The name is a combination of *Blue* (blue-violet laser) and *Ray* (optical ray). According to the Blu-ray Disc Association, the spelling of *Blu-ray* is not a mistake—the character "e" was intentionally left out so the term could be registered as a trademark.

BLU-RAY OVERVIEW

Blu-ray holds 25 GB on a single-layer disc and 50 GB on a dual-layer disc and has a data rate of 54 Mbps (compared to DVD's 9.8 Mbps). This means that both high-definition audio and video are possible at the same time. About 9 hours of high-definition video and up to 23 hours of standard-definition audio and video can be stored on a 50-GB dual-layer disc. BD players are backward compatible and will play standard DVDs and, in some cases, even upscale the picture to 1080p/1080i.

All Sony PlayStation 3 game units are shipped with a 2x BD drive. Stand-alone Blu-ray burners and discs are available, the price continues to drop, and pre-recorded titles are becoming widespread and readily available.

It's also possible that Blu-ray could allow the storage capacity to be increased to 100 GB to 200 GB in the future simply by adding more 25-GB layers to the discs.

BLU-RAY AUDIO SPECS

BD supports almost every audio codec format except MLP, but because of the high data rate, it's now possible to store up to eight channels of 96/24 LPCM audio and six channels of 192/24 without data compression and high-def picture. That being said, codecs, lossy and lossless, will probably be used if video is present on the disc.

Table 15.2 presents a list of the mandatory and optional audio formats. *Mandatory* means that all BD players are required to decode the format. A secondary audio track, if present, can use any of the mandatory formats or one of the optional codecs.

Table 15.2 Blu-Ray List of Codecs

Codec	Channels	Sample Rate	State	Type
Linear PCM (LPCM)	Up to 8	4	Mandatory	Lossless
Linear PCM	Up to 6	8	Mandatory	Lossless
Dolby Digital	5.1	2	Mandatory	Lossy
Dolby Digital EX	6.1	2	Optional	Lossy
Dolby Digital Plus (DD+)	7.1	2	Optional	Lossy
Dolby TrueHD	Up to 8	4	Optional	Lossless
Dolby TrueHD	Up to 6	8	Optional	Lossless
DTS Digital Surround	5.1	2	Mandatory	Lossy
DTS Digital Surround ES	6.1	2	Optional	Lossy
DTS Digital Surround 96/24	5.1	4	Optional	Lossy
DTS-HD High-Res Audio	Up to 8	4	Optional	Lossy
DTS-HD Master Audio	Up to 8	4	Optional	Lossless
DTS-HD Master Audio	Up to 6	8	Optional	Lossless

BLU-RAY VIDEO SPECS

Like HD-DVD, BD supports the video resolution of the DVD-Video standard, which is standard-definition 720×480i using the MPEG2 codec. The more efficient SMPTE VC-1 (which is based on Microsoft's Windows Media 9 codec) and MPEG4-AVC video high-def codecs allow resolutions of 720P, 1080i, and 1080p. Multiple codecs on a single title are possible.

BLU-RAY ADDITIONAL FEATURES

Although BD initially offered no additional features, version 1.1 (called *Profile 1.1*) will offer the same additional features as HD-DVD as an option—namely network connectivity, secondary video codecs, interactivity, and persistent storage.

BLU-RAY ADVANTAGES

- **Broad support.** More companies across a wide range of consumer electronics have come out in support of BD, both in pre-recorded content and recordable BD.

- **Holds more data.** Higher capacities means longer programs, more extras, secondary programs, and greater backup capacity in the BD-R.

- **Content.** More BD pre-recorded titles are available than for HD, although the difference is not substantial, and many titles are released in both formats.

- **Burner and recordable discs available.** As of this writing, BD burners and recordable discs were readily available and were sharply dropping in price.

BLU-RAY DISADVANTAGES

- **High manufacturing cost.** Because it's not based upon the DVD spec, BD requires a new replication setup that must be amortized through higher costs.

- **Difficult programming.** BD programming is based around a Java scripting language that requires a significant learning curve and a longer programming timeline to finish a disc.

Alternative Disc Technologies

Although it seems as if the current technologies will serve our high-definition needs for years to come, there are many other disc technologies on the horizon that may be able to offer a greater order of magnitude in storage, bandwidth, or both. Keep in mind that many of these technologies are only in development and may never see the commercial light of day. Among the alternative technologies are:

► **Holographic Versatile Disc (HVD).** This is a two-laser technology (blue-green and red) that, while totally incompatible with any of the current DVD-based technologies, promises an enormous capacity of up to 3.9 terabytes (3,900 GB) and a data rate of up to 125 Mbps.

- **Digital Multilayer Disc (DMD).** DMD is a clear optical disc that, unlike all commercial optical discs, has no metallic layers. It's based on a red-laser technology similar to DVD, which could allow the format to be manufactured in existing replication facilities with few modifications. DMD discs are composed of multiple data layers joined by a fluorescent material. The layers are coated with a chemical that reacts when a laser shines on it, and then generates a resulting signal. It is thought that 100-GB discs are possible for the format.

- **Enhanced Versatile Disc (EVD).** EVD is a format that's essentially a DVD, but that uses different and more efficient codecs, such as VP6 (from On2 Technologies) for video and Enhanced Audio Codec 2.0 (EAC) from Coding Technologies for audio. The Chinese government supports the development of the format, and they have announced their intention to officially switch from the current DVD by 2008 in an effort to decrease their dependency on foreign electronic products.

- **Forward Versatile Disc (FVD).** This is another red-laser format meant to be a less expensive alternative for high-def content. The specification calls for up to three 5-GB layers for a disc total of 15 GB using a WMV9 codec.

- **Polar High-Definition DVD (PH-DVD).** Based upon the current DVD spec and using a red laser, this optical disc promises a storage capacity of as much as 100 GB.

- **Ultra Density Optical (UDO).** This is a cartridge-based optical disc using both a blue-violet laser and phase change technology to provide a current capacity of 30 GB, although capacities of up to 500 GB have been theorized.

- **Versatile Multilayer Disc (VMD).** A high-capacity red-laser technology, the format reportedly can have up to four 5-GB layers for a current capacity of 20 GB, although eight- and ten-layer versions are supposedly in development.

Alternative Delivery Technologies

Shiny plastic optical discs, although sometimes a convenient means for backup and pre-recorded content alike, may eventually give way to online storage, on-demand content, and high-bandwidth fiber channel pipelines into our homes and businesses. If and when that happens, the current and future audio and video codecs will be more important than ever, since cable modem and DSL bandwidth still lag way behind what's available via an optical disc, such as BD or HD-DVD, and will stay that way for the foreseeable future.

Part III

The Interviews

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CHAPTER 16

About the Interviews

As always, the interview portion of the book is the most enjoyable from a personal standpoint. It's a wonderful thing to finally meet (at least over the phone) the people whose work I've been listening to for many years. Not only were the contributors most willing to share their working methods and techniques (something that mastering engineers as a whole are not known to do), but they were most gracious in taking time from their busy schedules to do so. For this I am most grateful and extend to them my heartfelt appreciation.

Since this book is about mastering as an entire profession, I've included a cross-section of the industry. Not only are the legends and greats represented, but also some engineers who deal in the specialty areas of mastering. Regardless of their perceived industry stature, they all toil in the everyday trenches of mastering, and much can be learned from their perspectives.

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CHAPTER 17

Interview: Greg Calbi

Greg Calbi started his career as a mastering engineer at the Record Plant New York in 1973 before moving over to Sterling Sound in 1976. After a brief stint at Masterdisk from 1994 to 1998, Greg returned to Sterling as an owner, where he remains today. Greg's credits are numerous, ranging from Bob Dylan to John Lennon, U2, David Bowie, Paul Simon, Paul McCartney, Blues Traveler, and Sarah McLachlan, among many, many others.

Do you have a philosophy on mastering?
GREG CALBI: I do. It really depends on the relationship with the person who brings me the tape. My philosophy in general is try to figure out how to improve what the person brings me, and then to try to figure out what his intent was. In other words, I don't just plug in my own idea without first really communicating with the client. It's a little tricky. It really is different for every project. You really have to get a good communication flow going, which sometimes is actually one of the most difficult parts of the job.

One time somebody said something to me that I thought was the best compliment that I ever got in mastering. He said, "The reason I like your work is because it sounds like what I did, only better." And that's kind of what I've always tried to do. I try not to change the mix; I just try to enhance it. I just go with the spirit of what was given to me, unless I really feel that it's totally missing the mark. And occasionally it does, because we're now in an era where you have a lot of people in the beginning of the learning curve because of the availability of the technology. People are getting into recording who have the resources because the cost of entry has gone down, but actually the qualifications for doing it have kind of diminished. The bar got lowered a little bit.

In terms of mixing?
GREG CALBI: In terms of being a recording engineer. I hate to sound like I'm criticizing the guys who have been getting into it, because I would do the same thing if I was young and musical. I'd buy the stuff and try to record at home and do a lot of what they try to do. But the fact is that they really haven't had the experience, so you get to the mastering stage now with a much greater need to augment what you've got, rather than the way it used to be back in the days when studios had staff engineers with an internship program. There was just a higher level of expertise that it took to get to the level where you could make a major record. You don't necessarily have that now, so we need to try to help them where we can.

Is there a difference between mastering from coast to coast or city to city?
GREG CALBI: There's really more of a difference from person to person. I've listened many years to all the different sounds that different guys have, and they really all do something different, and I respect every one of them for it. I could be blown away by something that any of 10 guys do, it's so recognizable.

We once hosted a great symposium that NARAS ran for their members. They had about 90 people come up, and the four of us—George Marino, Tom Coyne, Ted Jensen, and I—had the same mix to work on. We had 10 people in the room at a time, and we had a make-believe producer who was asking producer-type questions so people could see how a session went. We all EQed the same song and, after it was over, we all went out to the main room and listened to it with everybody there. All four sounded like four different mixes, and they all had their own thing about them. None of them sounded bad, but it was amazing how different they all were.

You really don't know what your own sound is. Maybe other people know and can identify with your sound more than you can, and every once in awhile someone comes in because they heard something on a record that you mastered and knows exactly what you tried to do with it. Even if it's only one person that picked up on it, it's just a great feeling.

Can you hear the final product in your head as you're running something down?
GREG CALBI: Yeah, I can hear where I want it to go. I use kind of an A/B method most of the time, so I'm always referring to other mixes on the album. What I try to do is get a listen to everything on the album before I start to work on it. That's something which I've started doing over the last two or three years, and now I almost do it religiously. I really want to know what the producer and the engineer are capable of doing at their best before I start to force it in a direction.

In other words, if the first song on the A side is like a nightmare, all of a sudden you're plugging things in and trying things and going back and forth, and you're just going crazy. You get into a certain negative mindset at that point where you think that this whole album is going to be tough. Then all of a sudden about an hour or two later, you find that all the stuff after that is starting to sound really good, and you realize that you might not have done your best work because you were forcing a mix into an area. Whereas if I go to the stuff that I really like hearing in the beginning, it gives me more of a realistic expectation of what I'm going to be able to get from this stuff later on. It's just a good way to give your ears something to compare to.

I even used to do it back before we had digital, where I'd cut a little piece onto the acetate behind me and go back and forth to listen. Every once in awhile there would be a real eye opener because it's a combination of ear fatigue and the way the mixes work where you think something is really working, but then all of a sudden your ears prick right up and you realize that you really didn't take it far enough. Or it could be the other way around, where you get a little ear fatigue and you start overhyping some things, and then you listen to what you know is good from earlier in the day, and all of a sudden that thing sounds nice and smooth, and the thing you're working on is starting to sound a little brittle. I use that method a lot to try to keep my ears fresh and to keep my aural memory locked in. It really helps me make the records cohesive from song to song, too.

Do you listen to the whole record before you start?
GREG CALBI: I'll listen to snatches of everything. I'll listen to maybe a minute or two of a few songs. You know the question I always ask? I'll say, "What's your favorite mix on the album? What's the one that everybody seems to really like?" because that'll give me an indication of what their expectation is. If they point me to something that I think is horrible and they think is great, then I know I have a combination of engineering and psychology because I need to bring them to where I know it might have to be. The funny thing is that as the years have gone on, they will throw it into my hands almost totally, and I have to drag them back into it. I find I work better when the client gets involved because when they take some responsibility for the project in the room, they'll also take that same responsibility when they're listening out of the room. A lot of mastering guys kick the people out and are really secretive about what they're doing, but I'm completely the opposite. The black magic thing is really totally overrated. It's kind of a fallback for a certain amount of not taking responsibility.

What do you think makes the difference between a really great mastering engineer and someone who's just competent?
GREG CALBI: Just a great musical set of ears. That's so important. I mean, there are some guys who just have a tremendous talent for creating something that's musically and aurally satisfying. But then the communication skill is another thing that makes somebody great, as well as a real good understanding of how to push the equipment and a willingness to try different things. It's kind of a combination of creativity and tenaciousness.

I think you have to really have a lot of pride in what you do. The aspect of pride is very, very important. There's no way that you can do this without being personally attached to the work. I always try to figure out whether this is an art or not. It's not really an art per se, but it has shared elements of what an artist does. You take possession of the thing.

How do you feel about the "level wars?"
GREG CALBI: It's gotten so insane. I'm a huge music fan and I listen to CDs constantly at home. I have to say that the CDs that always please me the most sonically are not the real hot ones when I bring them in here and look at them on the meters. I tell people, "If you want yours to be hot, I know how to do it, and I'll make it as hot as we can possibly make it and still be musical. But I just want to tell you that you may find that it's not as pleasing to you if you get it too hot."

The genre that I'm dealing with a lot, though, is not necessarily the genre where people really want to crank. I did something this week for Jay Beckenstein from Spyro Gyra. He's been around for 20 some-odd years, although I've never worked for him before, so I wanted to blow him away. I really wanted him to put this thing on and go, "Oh man, this guy's great." So I laid it on there pretty hot for him, and he calls me back and says, "I just want to tell you that this doesn't have to be the hottest record ever made. With this kind of music, it's really not that important." And I just thought, "Thank God this guy is not in that mode."

What do you think is the hardest thing for you to do?
GREG CALBI: Hard rock and metal have always been the hardest thing for me to make sound good because the density of the music requires a lot of aggressiveness. But what happens is, if the aggressiveness goes just that one step too far, it diminishes the music. You reach a point where all of a sudden it starts to reverse itself, where big becomes small and exciting becomes overbearing, and it works against the rhythms of the music. So you have to push it to the point, but if it's just one step past the point, it loses impact. It's a very weird phenomenon.

I've heard other people say exactly the same thing.
GREG CALBI: You go right off a cliff. There was one record that I did with a band called Reveille, which is a bunch of 16 and 17 year olds. It was really good, and I did everything I could to make it as loud as I could. What happened was this thing was put on a compilation with like maybe 13 or 14 other metal things, and, man, the other ones were so much louder. Some of them were terrible, but some of them were fantastic. It's the continuous puzzle of this trade, especially with that heavy kind of music.

I do a bunch of stuff that is more jazz and world music with kind of acoustic rhythm that's so powerful when it's nice and smooth because it's not so dependent on the level. But the metal and the hard rock are very, very dependent on it. If you catch it right, you've really created something really great.

How do you go about getting your level?
GREG CALBI: I wouldn't mind talking about it to a certain extent, but I'm still working on that all the time. What I do in general is try to use three or four different devices to a point where each one is just a little past the point of overload. I overdrive two, sometimes three and even four pieces of gear, one of them being an A-to-D converter, and the other ones being digital level controls. I find that if I spread the load out amongst a couple of different units and add them together, then I'm able to get it as loud as I can. I don't like to put soft limit or finalizing on things. What I find is there's a point where you're trading in rhythmic clarity and subtlety for loudness. I don't want to do that, although there are some types of music which do really lend themselves to it, particularly if a lot of the rhythm instruments have been sampled already, and the overtones have already been knocked off it. Again, it's pretty much content-based.

But I'll go back to what I said before, where a lot of times the things that seem the most powerful and the most pleasing in the home listening situation aren't necessarily the loudest ones. The loudest ones seem to be the ones that are the most blurry-sounding. Anybody who's working on trying to max their levels out has to see what happens to the strong dynamic elements when they start to get squashed.

I had a TC 5000 for awhile. I used to use it as a multi-band compressor, and I tried all kinds of different ways of getting that thing to max levels out. But if you take your original source tape and just forget about overloads and do an A/B at some peak level, I guarantee that you'll find that you've lost a whole bunch of depth, and that's a depth which people cannot recreate in their listening situation.

What's your signal chain like?
GREG CALBI: On the analog side, what I try to do is combine light and dark, solid state and tube. So I have a bunch of tube equipment. I have the EAR compressors and the EAR EQs—the MEQ and the regular one, like the old Pultec. And I have an Avalon compressor and an Avalon equalizer, which is a little bit more specific. Then I have something that we all have here at Sterling, which is a sum and difference box that was designed by Chris Muth that enables you to EQ and compress the center channel differently than the side channels. It's the most fantastic box; it almost eliminates the need for vocal upmixes because you can just EQ the center. You can also take sibilance away from the center without affecting the brightness of the guitars on the side, so you can really get pretty creative. I also have a Manley tube limiter compressor, one of those Vari-Mus, and one of Doug Sax's level amplifiers, which I'll use sometimes in between my console.

Occasionally, if something sounds really good, I'll just bypass my console and patch it directly into my A-to-D converter and use the analog machine as a level control. A lot of times with the DATs, I'll go into a Doug Sax line amp that I have to make them a little more analog if I don't need it to be EQed a lot.

I have an ATR analog deck with tube electronics and one with solid state electronics. I also have a Studer 820. Most of the time at the beginning of an analog session, I'll play it off each of those three machines and see which one sounds the best. I usually work with two different A-to-D converters. I have a dB Technologies converter, and I have one that the guys at JVC were fooling around with for awhile, which is excellent. I try to have two different converters at all times, one that maybe has a deeper bottom and better imaging and another one that's maybe a little more exciting in the midrange.

That's what I have on the analog side. The EAR compressors I also use as a level control. If you call me in a month from now, I'll probably have all different stuff. I don't buy a lot of gear, but I'm constantly changing what I'm doing and the order in which the gear gets plugged in. We use the Z-Sys for digital EQ. I have a Weiss compressor for digital compression, and Z-Sys has been fooling around with a compressor, which I also have.

I haven't had too much luck with digital compressors. With this Weiss thing I'm always trying to come up with something that works for everything, and every time I think I have a good preset and then try it on something else, it doesn't seem to work. I have probably the same stuff that a lot of the guys have. I think the sum and difference box gives us a little bit more of a chance at being a more creative.

As far as the person who might be trying to learn how to do his own mastering or to understand mastering in general, the main thing is that all you need is one experience of hearing somebody else master something. Your one experience at having it sound so incredibly different makes you then realize just how intricate mastering can be and just how much you could add to or subtract from a final mix.

I would also say to anybody who is trying to learn about mastering, realize that there's a hidden element that the more flexibility you have and the more time and patience you have, you can really come up with something that's going to be better. There's no shortcut to it. You just have to keep A-Bing back and forth and back and forth. It's pretty amazing how far off you can be sometimes, even when you think you're doing everything right. But then the satisfaction of knowing that you really got something great is just an amazing feeling.

How important is mono to you? Do you listen in mono at all?
GREG CALBI: I don't except to check for azimuth. I don't really work that way. I've had some clients who want to do it in mono, but it's not something that I do. I would imagine that there are guys who have fooled with it and really find that they do really great work that way because there's also guys that EQ a lot differently from channel to channel to get dimension and everything. I always feel by doing that you're taking balances in the mix and fooling around with them, and I'm very, very hesitant to do that unless an engineer comes and says to me, "You know, the guitar player made me push the guitar too far up on the right. Could you do something?"

I really don't want to give somebody something back and have them say, "What the heck did you do?" I just want them to listen to it and go, "Wow, it sounds better."

What are you using for monitors?
GREG CALBI: For six years it's ProAc Response 4. It's a big floor-standing model, almost like the size of a Dunlavy, but not as deep. I'm really happy with them. To me, they're well balanced and musical. They're powered with an Audio Research Stereo 300. I'm always fooling around with a whole bunch of crazy cables and with AC cords. There's a guy in LA doing some great AC cords for about $1,200 a shot.

Do you find it makes a difference?
GREG CALBI: I do blind tests with clients all the time, where I plug this cable into a converter or onto a machine, and they hear it right away. I'd like to buy like six more of them, but they're very expensive.

There's a guy over at Sony Mastering who apparently found that if he works between midnight and 8 a.m., there's so much less going on in the building that he thinks the power is better. You start getting crazy with stuff like this. It's only two tracks, so you take any little advantage that you can come up with.

Do you do your own production or do you have someone there to do it for you?
GREG CALBI: The production here is done in my room. I have an assistant who's a full-blown mastering engineer, but he works as my assistant and as a production guy in the studio. We get it to the point where the final EQ is approved, then we capture it as a 16-bit file in the Sonics. Once it's in there, then all the production engineers take it and make any 16-bit media that needs to come out, be it a DAT or a PMCD or CD or 1630.

Do you cut lacquers?
GREG CALBI: We actually have two lacquer rooms going pretty much all the time. We have a tremendous amount of cutting business because we do a lot of dance and rap music. I personally haven't cut a lacquer in six years, but I had 20 years of it before that.

Do you think cutting lacquers helped you in the way you work now?
GREG CALBI: There's nothing like cutting lacquers because of the attention that you have to pay to dynamics. It's so unforgiving. In terms of helping me, I think that you learn to concentrate on the dynamics because it's so critical to whether you're actually going to have a successful cut. You probably train yourself to see the VU meters and the music in one continuum. I think that it probably helped to focus me on how to concentrate on listening to music. Somebody today could say to me, "Did you like the way the song took off in the second bridge?" and I'd say, "I wasn't even listening to the structure of the song at this point. I'm listening to the whole." There's a whole other thing that's going on. There is a way that you listen to music when you have to cut a lacquer. You have to watch those meters and you have to make sure there are no hits that are going to make that record skip, so you're conscious of the rhythmic element.

See, that's the thing. There are guys who know how to make things sound really loud and big, but over-compression will keep the rhythm from working right. That's the thing that drives me nuts about the Finalizer and all this other stuff. Once you take away the beat, then you just don't have the same intensity any more. Maybe from cutting lacquer all those years, I started listening to drums a lot.

What makes your job easier? Is there something that your client can do to make things go faster, easier, better?
GREG CALBI: It's really common-sense stuff. One, stay off the phone: Let's get locked in and not have to constantly get our ears back up to speed. Two, know where everything is. Don't make me spend four hours rewinding tape because that's not really productive work. I always tell people that, as stupid as it may sound, probably the most important thing that they could do is just go into the session being organized so that they know where the mixes are. Three, be honest with the mastering engineer. Don't try to pretend that everybody likes something, and then later in the day start to reveal all the doubts that people had about certain aspects of the project. You'll just waste a tremendous amount of time that way. These are really basic human things.

What's the hardest thing you have to do in mastering? Is there a particular type of project that's harder than others?
GREG CALBI: I don't have any idea exactly why it happens, but the hardest ones are the ones that don't sound 100 percent, but yet you can't figure out what it is that could make it better. That's why I'm glad I have a lot of different things that I can plug in and just do signal path kind of stuff rather than EQ.

Another thing that's hard is when the low end is thin and light, because it's really hard to create low end when there is none. If you have a real muddy project, you can always clear stuff away and find something in there, but it's really tough when the bottom end isn't there. Most of the problem projects have to do with the bass being recorded poorly. If you made book of excuses, the chapter on bass would be eight times bigger than the chapter on everything else. He brought the wrong axe, we couldn't get another bass player, it was an acoustic bass, the room, the miking, the direct, the buzz, the hum.... It goes on and on.

But the fact of the matter is that you never have a great-sounding CD if you don't have a great bass sound. It can't be great unless the bass is great. It could be good, but bass is what takes it to the level where it's really something special. It's just a constant thing that you try to get to improve. It's the thing that engineers are the most frustrated about.

CHAPTER 18

Interview: David Cheppa

David Cheppa began cutting vinyl in 1974 and since that time has cut almost 22,000 sides. He is the founder of Better Quality Sound, which is currently one of the few remaining mastering houses dedicated strictly to mastering vinyl. Thanks to his intense interest and design engineering background, David has brought a medium once given up for dead to new, unsurpassed heights of quality.

Not too long ago, everyone thought that vinyl was dead, yet you're really, really busy.
DAVID CHEPPA: I don't think anybody else does as much vinyl cutting as we do. We do about 500 masters a month here, but only because that's the niche that it worked out to be. When things were waning back in the '80s, I was still acting like nothing had changed insofar as I was still looking for ways to develop and improve the medium.

You never think about vinyl being "improved."
DAVID CHEPPA: We've actually developed it quite a lot. In the old days, way, way back in the '50s, the first cutting systems weren't very powerful. They only had maybe 10 or 12 watts of power. Then, in the '60s, Neumann developed a system that brought it up to about 75 watts per channel, which was considered pretty cool. Then, in the '70s, the high-powered cutting systems came into being, which were about 500 watts. That was pretty much it for a while. I mean, it made no sense beyond that because the cutter heads really weren't designed to handle that kind of power anyway. Even the last cutting system that came off the line in about 1990 at Neumann in Berlin hadn't really had changed other than it had newer panels and prettier electronics. It wasn't really a big difference.

One of the things that I did was look for a way to keep the signal path simple and clean and free of anything that would affect the signal. I figure that a mastering engineer spent a lot of time and money to get it to where

he wanted, so I didn't want to alter the program when I finally got it. All I wanted to do was give them as faithful a reproduction as possible. What I went for was to keep the warmth of the vinyl, but have the power of the CD. But because we had CDs by then, nobody even cared about vinyl anymore. I mean, everyone in the cutting end was old school in their thinking in a lot of ways and didn't care much about improving the medium other than just trying to do what was always done. So using my background as a design engineer, I improved the cutting system, mainly the amplifiers. I pushed the power levels way beyond anything that we ever had.

In doing that, I sacrificed a number of cutter heads, and these cutter heads are about twenty grand apiece, if you can find one. In fact, Neumann doesn't really make them any more, but if you want them to build you one from scratch, they'll charge you $35,000 for it. If you can find one, you can pick it up somewhere between $10,000 and $15,000 right now, and maybe a burned-out one for about $5,000 or $6,000. It costs about $10,000 to repair it, just the way it is. Last year alone, I burned out four cutter heads to get everybody's product out the way I wanted. Nobody knows what we go through to get a really good faithful recording on the disk because when you master for CD, you don't usually master with vinyl ears. You master with an ear to whatever it is that you want and, as a result, you don't consider anything else.

When you get stuff in that doesn't use vinyl ears, what are the problems that occur?
DAVID CHEPPA: This is what I notice, and it's really the secret. The balance of the sound is the most important thing. You get a good mix where the elements are balanced well, and it cuts well as a result.

Frequency balanced?
DAVID CHEPPA: Yeah, in the sense of equalization, every aspect of it is balanced so that you don't have these anomalies poking out that you don't really want. It seems obvious that this is what you would strive for, but that's not what mastering guys generally do. They'll tweak things in all different directions.

I used to voice rooms to flatten out monitors so that they sounded good, and the way you get rid of all the problems is to feather any EQ that you used. The same with limiting and compressing. The best mastering I see is where people have feathered their work. It's almost like you're just fine-tuning. It's so subtle that you almost don't notice it. If it's a good mix, you can make a great master because the best masters have the best balance. It seems obvious, but it just bears out, especially in cutting.

Do you have to do a lot of mastering in the sense of having to do a lot of EQ and compression, or do you just do a lot of straight transfers?
DAVID CHEPPA: My goal is to take someone's work and keep it faithful and not touch it, but there are very few engineers that I don't have to do anything with their program. But my first approach is a subtle one. I'll do things where nobody even notices it because I don't want them to hear that I did anything.

The problem is taking something that's now in the digital domain and putting it in the physical realm. You're basically making that little stylus accelerate sometimes as much as 5,000 times the force of gravity, especially when you have program with a lot of percussive brilliance or sibilance sounds created by S's. The demands are so great.

And by the way, that's where all the power is required in cutting. In the physical world with sound systems, all the energy is in the low end. But in cutting, it's the exact opposite. All of the energy is in the upper spectrum, so everything from about 5,000 cycles up begins to require a great amount of energy. This is why our cutting systems are so powerful. One lathe has 3,600 watts of power, and our least powerful one is about 2,200 watts. It's devastating if something goes wrong at that power. If I get a master that's raw and hasn't been handled at all and there is something that just tweaks out of nowhere, it can take the cutter head out. So that's always a big concern.

If I'm not familiar with the material or the mastering engineer, then the first thing I'll do is dump it into our system here and look at what the sound spectrum is like to find out what kind of energy distribution exists. I can overview the entire project just at a glance and determine if there's anything that looks like it's going to be a problem. Unfortunately, it does take time, and it's not something I usually charge for.

We do everybody's work here, including all the major labels, but I treat every project as though I'm doing Babyface's album. Even when it's somebody's garage band, I'll give it the same care and interest because to me, every project is important. But that project may be a mess. If it's beyond anything I think I should be messing with, I'll call them and say, "Listen, this hasn't been pre-mastered for vinyl." "What do you mean by that?" "Well, there's percussive brilliance that's out of control." This is the problem in almost every case because sibilant distortion can occur on vinyl that doesn't occur anywhere else. It's because the velocities are so high and so quick that the person's playback stylus will literally chatter in the groove. That chattering sound seems to be a distortion, when in truth, the record might not have any distortion, but nobody can track it. I can actually cut records that nobody can track, which is useless.

The other problem with having the high power levels that we have today is that I have to figure out what kind of client this is going to and what kind of turntable and cartridge he'll be using. My lab turntable uses a high-compliance cartridge, but that isn't what they're using in a club. If they're going to use a DJ setup, let's make it so they can play it. So that's another consideration.

Where does most of the vinyl go?
DAVID CHEPPA: Today there are so many markets. The DJ market, or the dance/rap/hip-hop market, is probably the greatest number. I think 80 percent of it goes there. The other percentage is really only a few percent, like classical music. We're having a resurgence of swing music and big band that's incredible, and a lot of music that we're re-mastering was done in the '60s and '70s. Everything that Polygram ever did and everything that Motown ever did, they're being re-mastered, and we're re-cutting them.

We're actually getting a better record now than they had back then because you're hearing things that they couldn't hear on the original masters. Also, the cutting systems weren't that evolved back then, either. Everything's been improved so much.

What else has improved?
DAVID CHEPPA: One of the things that people used to do is compress and limit and EQ to try to make it go to vinyl. My goal is to take whatever the person had and make it go to vinyl without going through anything. That's a real feat at times because, again, with a master that was prepared digitally, people don't think there are any limits. They do whatever they do to make it good for CD. I try to keep a straight path from whatever master machine I'm working from, whether it's an analog or a digital source.

That's a big task for me because some things are not physically possible. I'll get masters that I can't cut, and the reason is they're so rich in harmonics in the upper spectrum, which you can't even hear.

Because it's so distorted or squashed?
DAVID CHEPPA: What's happened is it's almost limitless in the way you can control the sound now, where the equipment in the earlier days wouldn't handle the frequency or transient response or the power levels. Most of the gear today is much more responsive. When people EQ, they don't realize that they may be adding harmonics that they're not hearing. Where something like a flute's highest fundamental frequency may be just under 5,000 cycles, its harmonics go out to 15,000, 18,000 cycles, and beyond.

My biggest challenge is that they're EQing this top end so that it sounds crisp and nice, but they don't realize that things like bells and cymbals are adding harmonics that they're not hearing and that may make it impossible to cut. I'll try to tame that portion of the sound spectrum that they can't hear in the first place because it won't go to vinyl otherwise.

A lot of guys who are cutting today can't figure out why they're having trouble, so they just back off on the level or they smash it or just EQ it all out. The only problem with that is you then affect the brilliance and the air and the transparency. So sometimes I'll go in and I'll just tailor those harmonics.

What is the master format that you usually get in?
DAVID CHEPPA: I get everything, but most of the stuff comes on optical, like a CD-R. The reason I prefer that is—and I don't care what anyone says—it's the most stable format we have right now. I would always prefer it if someone can give me an optical format because I know, no matter where it was burnt, unless their burners are bad or they have a defective CD, it will always work.

Do you load it into a DAW?
DAVID CHEPPA: We're using several systems here. Some of my cutting is done off a hard drive so I can assemble something quickly if you send it to me out of order. That happens a lot. I may actually do some EQing in there if I notice something. I'll maybe taper the high end a little bit, or if there are sibilant problems, I'll do some de-essing. Again, I don't like doing any of this stuff because it affects the program as far as I'm concerned, but I'll try to be so subtle and feather it.

A lot of times I will cut a little test on the outer diameter of the record. Not the area that we're sending for processing, but an area that I can play with. I will do that until I make sure that whatever is done is faithful to the original master, because there's so many variables in cutting that the response can change drastically by the stylus temperature, the stylus being dull, even the temperature in the room if the room is very cold and the lacquer is cold. I might turn up the temperature on the styli. The higher the temperature of the styli, by the way, the more resolution you can get. If you increase the temperature a little bit, it will cut more easily and maintain the response. But I only run styli for a few sides or a couple of hours total and then I discard them, because I try to maintain a certain standard. As soon as they get dull, then the response goes way down, and that's not good.

With a lot of rap and hip-hop, do you have problems with the low end?
DAVID CHEPPA: The answer is yes and no. It's almost always no good if they haven't really mastered it because the kick may be boosted so severely that there's no way that you can get any apparent volume.

That's the other thing that I try to do—get the most apparent volume I can get on the medium. I had a Sublime record that I was cutting last year, and the sides were 28 minutes, which is just too long. The longest side I ever cut was somewhere around 35 minutes, and that was a spoken-word record. But what I did was alter the EQ just a little bit to give them a sense of volume where there really wasn't one. Again, I think I did it in a way where nobody knew, but the result was okay. It was kind of a compromise, but there are so many compromises you have to make sometimes, you just don't want them to be noticeable.

So what's your signal path then?
DAVID CHEPPA: The signal path is direct. I mean, once I go out of the converters, I'm going right into the cutting system. Sometimes I'll cut off the converters, and sometimes I'll cut right off the analog source, but I try to avoid anything that's going to alter that signal path at all. That's where I have the danger of destruction on the cut because of the power levels, because if there's a high frequency that's not controlled, the cutter head can't dissipate the heat fast enough, and it's going to blow.

Normally you'd go through a limiter/compressor, maybe some kind of EQ, all kinds of amplifiers and transformers. I've eliminated everything. In fact, I even went through and pulled all the transformers out of all the equipment because I didn't want the changes that occur from the transformers. Most guys that cut around the country still have older systems, and because they've accepted the way things have been for so long, nobody thinks about it. But the signal path is so blocked with things that actually kind of blur the original source a little bit, and they don't even know it.

Another thing that I do and no one else does is run my helium pressure—used to cool the cutter head—seven or eight times of what is normally used because of the power that I now have. Because if I don't cool that cutter head down, I know I'm going to lose it. I found that I was able to cut higher levels with more high frequency that way. The factory settings work, but they never intended their cutting systems to be pushed as hard as we push them.

It really must take a lot of experience to cut a good record.
DAVID CHEPPA: If you just want to cut a mediocre record, you don't need to know a lot of anything. If you want to cut a better record, it's good to know something. If you want to cut an incredible record, you need to have an understanding of the physical world and the physical laws that govern it. You have to know what the limits really are, physically and electronically. So I think it's a balance of art, science, and technology.

How many sides do you cut a day?
DAVID CHEPPA: Some days I'll do maybe 25 or 30 masters. That's pushing it and about the most that I can do. Now, if they're short, I can do more. I have some that are 25-minute sides, so they take a half-hour to cut, but sometimes the preparations are pretty hard. Like when I was doing those Sublime masters—because I wanted to get it loud, I spent hours preparing for something that was only going to take a half-hour to cut on each side. But on the short side, on dance records like the 7" singles, I may be able to do four an hour or sometimes even more.

That's assuming that you don't have to do any fixes.
DAVID CHEPPA: Yeah, like I said, we can do pre-mastering here, but I usually reserve that for fixing problems. I figure I'm going to stick with what we do best. We cut here and we'll do pre-mastering when we need to, but I don't want to compete with the people that supply us with masters. My goal is to give them something beyond anything they expected on vinyl. In other words, whatever it takes to get this guy's record to sound incredible, that's what I want to do.

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CHAPTER 19

Interview: Dave Collins

A mainstay at Hollywood's A&M Mastering for many years, Dave Collins has brought his unique approach to a host of clients, such as Sting, Madonna, Bruce Springsteen, and Soundgarden.

What is your philosophy on mastering?
DAVE COLLINS: The first philosophy is like the Hippocratic oath: Do no harm. The client is investing a tremendous amount of trust in the mastering engineer when he gives you the tape and expects it to sound better than it did when he brought it to you. I personally think experience is as valuable as equipment in a large sense because after you've done it for 10 or 20 years, you've heard almost everything that can possibly go wrong and go right on a mix. So you can, in one respect, quickly address people's problems.

Today we are in kind of a funny situation because the definition of mastering has become a little diluted, in my opinion. An L1 plug-in does not a mastering engineer make. Just because it says "Mastering" on the box and there is a preset called "Rock & Roll" in it, that's not what it's all about. When a guy writes a book, he doesn't edit the book himself. He sends it off to an editor, and the editor reads it with a fresh set of eyes, just like a mastering engineer hears it with a fresh set of ears.

Every so often I'll have a client that I work with all the time, and his budget is gone by the time he's ready to master. And so he says, "Well, I'll go in the studio and I'll hook up a Massenburg EQ to my two-track and I'll do a little equalization, and I'll put a compressor of some type on the output of it." But he'll ultimately call back and say, "Well, I don't know what I'm doing here. I'm just making it sound worse."

And that's kind of analogous to some guy trying to edit his own writing. It is the impartial ear that you get from your mastering engineer that is valuable. All this equipment and new technology that we've got is a great thing, but you're really asking for someone who has never heard the record before to hear it for the first time fresh.

When I listen to a record I've never heard before, I don't know that the guitar player was fighting with the singer through the whole session, or everyone hated each other by the time the record was done, or whatever political bullshit entered into the equation. I just listen to the sound that comes out of the speakers and take it from there.

What distinguishes a great mastering engineer from someone who is merely good or competent?
DAVE COLLINS: It's probably two things. The best mastering engineers have a sensibility to the widest range of music. And I think some mastering engineers get kind of pigeonholed into a certain style of music—"Oh, you've got to take your rap record to studio X and you have got to take your guitar/pop record to studio Y"—and I don't really subscribe to that. I think the best mastering engineers understand a wide range of music. Believe me, I buy tons of CDs and listen to everything so I can stay current with what is going on because I have got to get what the fans are hearing and understand that. So having aesthetics for a wide range of music is probably a fundamental skill.

Secondly, I would say that having a technical background, especially these days, certainly doesn't hurt because both recording and mastering now are far more complicated than ever before. The palate of signal processing that you have today is enormous, both in analog and digital, and it is growing all the time. Unfortunately, only one percent of all the gear that is out there is really optimized for mastering. Mastering is a really small market, and only a couple of companies really build stuff for mastering. TC [Electronic] will tell you that some of these boxes are made just for mastering, but they really aren't.

Yeah, you don't see many pieces.
DAVE COLLINS: I actually use one of their boxes, called the dB Max, which is designed to be a radio station processor and is not even designed for mastering. If you spend a little time fooling around with it, it actually works great.

What does it do?
DAVE COLLINS: It's a great de-esser and it's a really good limiter. The Waves L2 is a better peak limiter, but we still use this TC box, which has a million different functions, just for de-essing. People kind of look at it

sideways when they come in because it sort of looks like a Finalizer, but it's a good box.

It does some other things that are handy too. It will make compatible mono, if you have a mix that doesn't sum to mono properly. It has a 90-degree phase shift that you can introduce to the signal that will stop elements of the mix from canceling out in mono. I have done that when we've sent stuff overseas for music videos, where stereo TV audio is not as popular as it is in the US. So making a good compatible mono program is sometimes useful.

How important is mono to you, and do you listen that way often?
DAVE COLLINS: I always check in mono, and I think mono is very important. One thing that is overlooked sometimes is the fact that the signal on your FM radio becomes increasingly mono as the signal strength decreases, so it is important to check mono.

One thing that happens after I've listened for a long time, I can tell by how phasey it sounds to me in stereo if it's going to sum to mono. And once I get a certain amount of that eyes-sort-of-crossing feeling, I can pretty much tell that it is not going to sum to mono. But yes, we always check for compatibility. I've certainly had mixers come in with stuff, and I'd say, "Man, that is some wide-ass stereo you got going there. How does it sound in mono?" And the guy goes, "I don't know. How does it sound in mono?" And of course you put it in mono, and now one of the guitars has disappeared. So, it's an issue, but probably less important as time goes on. But I think it's still significant.

Can you hear the final product in your head when you first run through a song? Do you know where you are going with it before you go there?
DAVE COLLINS: No, not always. And in fact, I frequently go down a dead end EQ or processing wise. There are some styles of music that I will intrinsically get faster because the sonic presentation is pretty standardized in a lot of ways. So, there are times when I can hear 90 percent of what it is ultimately going to sound like immediately when I put it up, and there are other times when you go around in a big circle.

I guess when we were talking about the philosophy of mastering, what I should have added was, one of the hardest things—and it took me forever to get this—is knowing when to *not* do anything and leave the tape alone. As I have gained more experience, I am more likely to not EQ the tape or just do tiny, tiny amounts of equalization. I think some people feel like they really have to get in there and do something. They really have to put their stamp on the tape somehow.

I don't really care about that. I only care that the client is happy and he comes back. I don't really feel that I need to put any particular personality on it. And hey, if the tape sounds good, let it sound good. To backtrack on the whole philosophical aspect, I am a fanatic about being able to reproduce the master tape properly. I've built an entirely custom analog tape playback system to get every bit of information and music off the client's tape to begin with, and what I have found is as I optimized that system, I have to EQ less. The music will require less EQ as you improve your chain.

Sometimes people are fighting their own electronics. They have a piece of gear in the signal path that sounds dark, for instance. Now, suddenly, you must compensate with equalization at some other place. As I got my system dialed in, I found that I EQ less than I did 10 years ago.

What is your signal chain?
DAVE COLLINS: I used to work in electronics before I got into audio, so I had some background in analog engineering. It started by finding things that sounded good—like say an ATR-100, which everybody likes—and doing some modifications and optimizations of the circuitry. Some of these are due to the fact that when the ATR was built, some of these components and technologies just didn't exist in the mid '70s, and today they do. So we can bring some of it up to date.

The tape playback system that I use now is a half-tube, half-transistor system that sounds great. I have had a lot of people come in and really be surprised at what was on their tape that they didn't hear in the studio because you're reproducing it in a much more resolute, much more accurate way. More often than not they're hearing things that they like that they didn't hear in the studio.

But my philosophy is, optimize every inch of the chain and really get it as clean and as pure as you can, because you can always screw it up some other way. You can always distort it or do whatever you want to do. But if you don't start with something that is clean and transparent, that always hampers you. You have to begin there.

Do you do many analog versus digital shootouts?
DAVE COLLINS: Not many, but when a client mixes to 1/2" and DAW and he brings in both, and we do a very careful level-matched A/B between the two sources, whenever the DAW wins, it is because they couldn't set up their 1/2" machine right, in my opinion. A properly aligned 1/2" machine should always be a given, but these days it's kind of a lost art. I get stuff where the azimuth is on the moon, and they obviously haven't put an MRL tape up on the machine for 20 years.

Are you getting anything in that is recorded at a higher sample rate?
DAVE COLLINS: Yes, it sounds terrific, but when we do blind tests on 48k versus 96k, no one can consistently hear the difference. Everyone loves the sound of 96k when you're sitting there and you know what position the switch is in, but—at least in our tests, which used analog tape as the source—no one could consistently tell whether it was 88.2 or 44.1.

I'm not convinced either.
DAVE COLLINS: Well, it's technologically a funny question because I guarantee if you like 96k better, it is not because you are hearing to 48k. Our hearing has not evolved another octave of range just because 96k is being marketed. What it does do, and it is kind of an arcane technical point, is relax the anti-alias and anti-imaging filter requirements by half so you need half as much filtering at 96k for the same bandwidth. But to me these tests are a little hard unless you had a band set up live on the floor and took the signal right off a mic preamp or something like that. I'm sure that would be a more accurate test of 96k. I mean, when we use 1/2", I can see that there is some slight ultrasonic information present on the tape. But so far as we've been able to tell, I don't really hear any significant difference.

The average person is not going to hear it.
DAVE COLLINS: That is something that we definitely have cried in our beer about because my mom can tell the difference between stereo and 5.1, but I can get a room full of professional audio engineers, and we can barely hear the difference between 44 and 88k. So you have to be careful from a marketing point of view where this stuff goes because the audiophile market is like one tenth of one percent of the total audio sold, and it's a strange world to be in. I'd rather present compelling multichannel stuff at 44k.

I really wanted to like 96k because, from a technical point of view, there are some interesting things that can be done with it, and it just gives you twice as much room to work from a processing point of view. But when we tried to do blind tests—I've done it twice now, once with the Prism gear and recently with the dB Technologies gear—the results were statistically about the same as flipping a coin.

What converters are you using?
DAVE COLLINS: We're using dB Technology A/Ds, and for 96k I'm using the dB Technology D/A. For 44.1 I'm using one I built myself based on Ultra Analog components. We just went through a big shootout of all these converters and tried the HDCD, DCS, Prism, dB, and Mytech. It's funny, the Prism and the dB Technologies sound almost identical. I mean, we were just pulling our remaining hair out to hear the difference. But ultimately, when you compare it to the 1/2" tape, the dB was ever so

slightly closer to the master tape. If I didn't have the master source to compare to, I would not have been able to tell you one was better than the other. If somebody just gave me a CD that had two tracks on it, and I didn't have the master to refer back to, I could not have told you. They are both good products.

What is the hardest thing that you have to do? Is there one type of operation or music that is particularly difficult for you?
DAVE COLLINS: Well, the hardest thing to do is a compilation album. These "Very Special Christmas" albums are a good example, where you have 13 songs with 13 producers and 13 engineers and, in some cases, 10 different mix formats. Those are the hardest, just from a strictly sonic point of view, to try to get any consistency to.

Second to that is working on projects that have a "too many cooks and not enough chefs" condition, where you've got a lot of people kind of breathing down your neck and a lot of people with different, usually contradictory, opinions. Some of those projects—and usually they are your major-name artists—can be a little problematic because you have so much input and everyone is trying to pull you in a different direction at once, so that can be a little nerve wracking. But it's all in a day's work.

What do you enjoy the most?
DAVE COLLINS: The day after the session, when the client calls and tells you everything sounds great and, "I can't believe how good my CD sounds. I had no idea my mixes sounded that good." Seriously, they do come. That's the best, when I have someone who really got what I was doing and really got what my room is able to produce. It's not every session, of course, but those are good calls to get.

What are you using for monitors?
DAVE COLLINS: Presently I'm using Genesis 500s for the mains and Quested 108s for the minis. The mains are soon to be changed to B&W 802.

They seem to be popular these days.
DAVE COLLINS: It's a good speaker. I never liked the old B&W 801s. This new one is really amazing. I don't find much to criticize in it other than it is bloody expensive.

The 801s seem to be the classical standard, both for recording and for mastering.
DAVE COLLINS: They were. I've heard them many places and I never really understood why. It's like saying my car only turns right. What good is a speaker that only works on classical music? That means it's not accurate. You mean it won't play a kick drum?

Tell me more about your signal chain.
DAVE COLLINS: The analog signal path is a Studer 820 used just as a transport. We use a Flux-Magnetics playback head that's connected to the outboard tape playback electronics that we talked about before that is a half-tube, half–solid state. That feeds an all-custom analog console. Basically, the tape machine feeds some passive attenuation, and from there I've got a custom EQ that we use.

I've got a Prism analog EQ, a Manley Variable-Mu compressor, and a heavily modified SSL console compressor.

That thing is great. I was just telling somebody at lunch today that if you take a Manley Vari-Mu and an SSL compressor and have those in your console, that covers an enormous range of dynamic possibilities. You've got the kind of in-your-face nervous sound that an SSL can give you, which is something that people respond to very well, and then you've got the Manley, which is much more polite. The Manley has some sort of magic features to it; just running stuff through it sounds good. It is probably phase shift and distortion, but it sounds good. And we've got a Waves L2 limiter (serial number 0) and a dB Technology A/D converter. I also use that TC dB Max that we discussed.

Basically what I do is A/D convert the output of the console, and then from there we'll do maybe a tiny bit of EQ. I've got one of those Weiss digital EQs, which is a wonderful box, but to me, if you've got good analog EQ, it's really hard to beat it digitally. But sometimes for a few touchups here and there, I think it's very valuable.

As far as limiting, a digital limiter is just far superior to any analog limiter. You just can't get analog to do the things you can do in digital. And with today's kind of stupid dB level war that you have to fight, you're just skirting the hairy edge of distortion every step of the way. I mean, to get a CD to the level of the loudest CDs today, it really requires kind of tiptoeing around distortion.

I never would've thought that we would be cutting CDs at this level. It's to the point where a large amount of our day is optimizing the gain structure in the console and checking what kind of limiter you're going to use and how you're going to use it just to get the CD as loud as you possibly can. I don't get it. I have to play the game because if you want to stay in business, you've got to compete on absolute level, but it's really a horrible trend. I wish all mastering engineers would speak out about this because it sucks.

I buy records that I really want to listen to, and they are so fatiguing. It's impossible to get that amount of density and volume on a CD and not make you want to turn it off after three songs. I don't know how to put it in print in a diplomatic way, but when you get mastering engineers together and you get a couple of beers in them, they'll all agree that CDs are too loud. We hate it and wish we didn't have to do it, then it's right back to work on Monday and squeeze the shit out of it all over again.

Part of the problem is everything gets squeezed to death even before you get it.
DAVE COLLINS: I have a client that says before he sends the client home with a CD-R, he has to run it through some kind of compressor, limiter, Finalizer, you name it, just for their take-home copy, or the artist doesn't respond to it.

My joke about this is the whole problem started when they came out with multi-disc CD changers. Because before, by the time you took the one CD out and put the new CD in, you forgot what the volume was on the last one. If you had to adjust the volume control, no problem. But now when you've got the six-disc changer, one CD comes on and it's 10 dB quieter than the last one, and this next one comes on and it blows your head off, it's a problem. I don't know what the answer is. The frightening part to me is when we're right at the threshold of a 24-bit home format, we're still probably going to squeeze it into the top of its dynamic range. I hope we don't because I would love to hear some of these new DVD audio releases actually using the available dynamic range. Nobody uses any of the available 16-bit dynamic range as it is.

In mixing, if you don't squash it, the client isn't happy.
DAVE COLLINS: It's true. And believe me, it's the same way in mastering. When I get it to where I'm almost uncomfortable with the amount of processing I'm doing, the client responds to it and loves it.

Do you cut lacquers?
DAVE COLLINS: We still have one lathe set up. Every year we get together and say, "Well, this will be the year when we pack the lathe up and sell it or put it in storage." And every year there's just a little bit more work than the last, and it's frankly enough to keep us in business with lacquers.

Lacquers are funny. You have three types of clients. You've got the guy who can't afford to make CDs and can press a white-label 45 for 30 cents. You've got the total high-end boutique client who wants to put out 50,000 copies of his new record on vinyl because it's cool. And then you've got a DJ who just wants to take a 12" lacquer to play in a club. They bring in a CD, and you basically give them a flat constant pitch transfer to a lacquer so they can scratch on it in a club.

Do you cut yourself?
DAVE COLLINS: I have, but not really. I have to say, cutting is really fun in a sense because it's a skill. Cutting a loud record is very difficult and it requires an enormous balancing act of physics and sonics. Any idiot can make a loud CD, but not any idiot can make a loud record. And in a way, I miss it a little bit. But I guess I really don't because all the physical limitations of a record are gone on the CD, and nobody ever worries about the laser jumping out of the groove.

Do you ever have to use effects? Anybody ever ask you to add reverb?
DAVE COLLINS: Oh, sure. We've done a lot of soundtrack mastering at A&M, and it's very common to add a touch of reverb at the final stage. Generally, you won't want to add reverb to a whole pop mix because it gets too washy. But five times a year, I bring up an Eventide DSP4000 because I want to flange the whole mix like you hear on that Lenny Kravitz track, where the whole thing goes through a flange and you cut it back into the regular track. And sometimes we'll go to the telephone limited bandwidth kind of sound for a measure or two and back again, or something like that.

But generally speaking, I hope that by the time the record gets to mastering it doesn't need effects. But I've done things like overdubbed vocals in the mastering room before. I've overdubbed guitar solos in the mastering room too. Live, right to the master. I remember the last time we were doing vocals, the guy was like, "So, what kind of cue mix are you gonna send me?" I said, "I'm gonna turn the level down low on these speakers, and you can listen to it and you're gonna sing. How's that?" It does happen, but fortunately not often.

When you have to add effects, what box do you use?
DAVE COLLINS: Well, for reverb, I like the old Lexicon 300. I think if you get into the parameters on that thing and spend some time with it, it's really a good box. For general purpose, I think that Eventide Orville has just got some great programs in it. Whenever I need to flange something or add some weird slap-back to a section or something, I always reach for that because it's got a digital I/O.

Do you ever have to do something where somebody cuts the heads or tails off and you have to fix it?
DAVE COLLINS: Oh yes, sometimes you'll have to add a little reverb at the end just to give you something to fade over. I generally try to caution people, don't trim it too tight because it's a lot easier to take it off than it is to put it back.

What do you think the mastering house of the future is going to look like?
DAVE COLLINS: I think it'll look fundamentally the same as it has always looked because the basic requirement for accurate monitoring in an accurate acoustical space will never change. It will always have recognizable elements of it.

The mastering house of the future will have at least five loudspeakers. The mastering house of the future will have much more digital processing, and there will be a much wider palette of digital processing to choose from. I'm sure you're going to walk in, and it's going to look like the bridge of the Enterprise, but the basic requirements of good acoustics and good monitoring will always be there. That's one thing that will always stay the same.

CHAPTER 20

Interview: Bernie Grundman

One of the most widely respected names in the recording industry, Bernie Grundman has mastered literally hundreds of platinum and gold albums, including some of the most successful landmark recordings of all time, such as Michael Jackson's *Thriller*, Steely Dan's *Aja*, and Carole King's *Tapestry*. A mainstay at A&M records for 15 years before starting his own facility (Bernie Grundman Mastering) in 1984, Bernie is certainly one of the most celebrated mastering engineers of our time.

Do you have a philosophy on mastering?
BERNIE GRUNDMAN: Well, I think that mastering is a way of maximizing music to make it more effective for the listener, as well as maybe maximizing it in a competitive way for the industry. It's the final creative step and the last chance to do any modifications that might take the song to the next level.

There are a couple of factors that come into play when we're trying to determine how to master a recording. Most people need a mastering engineer to bring a certain amount of objectivity to their mix, plus a certain amount of experience. If you [the mastering engineer] have been in the business awhile, you've listened to a lot of material and you've probably heard what really great recordings of any type of music sound like. So, in your mind, you immediately compare it to the best ones you've ever heard. You know—the ones that really got you excited and created the kind of effect that producers are looking for. If it doesn't meet that ideal, you try to manipulate the sound in such a way as to make it as exciting and effective a musical experience as you've ever had with that kind of music.

Now, you can only go so far. Mastering has certain limitations. You can't completely change the mix, but you can certainly affect it a lot. Sometimes you can affect it dramatically—so much that it really becomes much more engaging musically for the listener. And if somebody brings in

something that's better than what you've heard, you have to be open enough and sensitive enough to let that music affect you. So you have to really be willing to admit sometimes that, "Hey, this is actually better than anything I have ever heard before." All it means is that you have a new ideal.

So, I think one of my biggest philosophies is that the music really has to tell you where to go. What that monitor is telling you is the truth, as long as you have a good monitor. You manipulate the song in one direction and you go, "No. Now the music is aggravating me. I'm not getting as good an experience." Instead of the things that are supposed to contribute to the effectiveness of the music, you're hearing all the elements of the mix getting obscured and muddy when you're manipulating the sound. You have to be aware of that, and be aware of the elements that are important to make that thing effective. It's one of those back-and-forth kind of things.

In the end you really have to be sensitive to whether you're really making it better, rather than just some intellectual pursuit where it's as bright or as loud as somebody else's. That's not really a great criterion for a musical experience. The real question is whether it's really communicating better musically? Emotionally? And I think that's something that all mastering engineers struggle to open themselves up to—whether or not this manipulation is really going in the direction that's beneficial for the product.

What about the interaction with the client?
BERNIE GRUNDMAN: Yes, you have to interface with the producer or the artist too, because they might have a vision that may be slightly different than where you intuitively want to take it. They might want to emphasize some aspect of the music that you may not have noticed. So a lot of it is definitely trial and error on your part, but it's also give and take between the producer and the artist because you can't sit there and arrogantly think that you know where this recording ought to go and that they don't.

Not that you shouldn't suggest things, because more often than not, the producer will say, "Yeah, I like where you're going with it. You're making it better than it ever was." Hopefully you get that kind of response. And then sometimes they'll have comments like, "Yeah, I like that part, but it's hurting this other part of the music. When you're pushing it here, it's hurting it over there." Or, "This is an element that I don't want to lose." It's all a learning process. I always say that we're all trying to get to the same place, but we're just trying to figure out how to get there. We want to get the best musical experience and be competitive.

So we've got all of these aspects that we're kind of struggling to maximize, and sometimes it takes two or three passes before it's right. They take it home, listen to it, and say, "No, let's try to get a little more of this out."

Or, "Can we do this or that?" You try to do the best you can, but mastering is usually a little bit of a compromise in a lot of cases.

Can you hear the final product in your head when you first run something down?
BERNIE GRUNDMAN: Well, you do get ideas. If you've been in it awhile and you've heard a lot of things, then you know where to go. Like if you put on a rap record, you know that it's very rhythm-oriented and it has to be really snappy and punchy on the bottom end. You know that some of the elements are really important and that this kind of music seems to feel better if it has them.

Or they may have had a monitoring system that had a lot of bottom end, and the tape comes out bottom-light as a result, but they thought they had it right. That's why probably the single most important piece of equipment that a mastering engineer can have is his monitor, and he has to understand that monitor and really know when it's where it should be. If you know the monitor and you've lived with it for a long time, then you're probably going to be able to make good recordings. The only problem with that is, if the monitor is something that is a little bit esoteric and only you understand it, it's very insecure for the producer or the artist because they don't think it's there, and you have to reassure them all the time. That happened to me when I first worked at A&M and I had a monitor system where I knew what it should sound like, but it was really kind of wrong for everyone else. They had to trust me—and they did, but I could see them get really insecure and concerned. So in my studio I've gone to great lengths to make it a very neutral system that everyone can relate to.

What are you using?
BERNIE GRUNDMAN: We put it together ourselves. We build our own boxes and crossovers, and we use all Tannoy components. We have it all mixed in with different elements that we feel are going to give us the best sound. It's not that we're going for the biggest or the most powerful sound; we're going for neutral because we really want to hear how one tune compares to the other in an album. We want to hear what we're doing when we add just a half dB at 5k or 10k. A lot of speakers nowadays have a lot of coloration and they're kind of fun to listen to, but boy, it's hard to hear those subtle little differences. We just use a two-way speaker system with just one woofer and one tweeter so it really puts us in between near-fields and big soffited monitors.

Do you use only that one set or do you use near-fields as well?
BERNIE GRUNDMAN: We have some NS10s and some little Radio Shack cubes. These are things that a lot of people around town like to hear what it's going to sound like on. Usually if you can get it sounding good on our main system, it's just that much better on the other ones.

When you're processing, are you doing that prior to going into the workstation? Are you doing that in the analog or the digital domain?
BERNIE GRUNDMAN: We do a lot of our processing analog. A lot of times we'll put it right on the computer already EQed and processed. Sometimes we don't. It depends on the project. Some of the stuff I'll put on the computer and then I'll run it through the board.

Is your console custom built?
BERNIE GRUNDMAN: Yeah, we build our own equipment. It's built mostly as an integrated system to avoid a lot of extra electronics and isolation devices and so forth. When you buy most pieces of audio equipment, each one has its own isolation transformers or electronically balanced outputs, or however they arrive at a balanced output. But when we buy outboard equipment, we completely rebuild it and put all of our own line amps in and take out the transformers or the active transformers. You'd be amazed at how much better they sound as a result.

We have all separate power to each one of our rooms and a very elaborate grounding setup, and we've proven to ourselves that it helps time and time again. We have all custom wire in the console. We build our own power supplies as well as everything else—the equalizers, everything.

It must take a long time.
BERNIE GRUNDMAN: Yes, it takes about three to four months to build a console. Sometimes six months. We built one for our studio in Japan that's a 5.1 six-channel board. We had to design it specially so that we could go from two-channel or six-channel with just a push of a button.

Are you going to do surround sound?
BERNIE GRUNDMAN: Japan is running 5.1 just for DVD-Video. We have a room that's designated for it here, and we're building a second six-channel board that will go in there.

What are you going to use for subwoofers?
BERNIE GRUNDMAN: We're using two Vandersteins, one on either corner of the room up front on either side. The five main channels are all full-range speakers.

Do you still cut lacquer?
BERNIE GRUNDMAN: Oh yes, we sure do. In fact, I'm going to be cutting lacquers all afternoon. We have one room where we cut all of our lacquers now. We used to have lathes in every room in our old studio, but we figured there would be less vinyl work in the future, so now we have just one room that has two lathes in it.

One of the lathes is for the audiophile guys and it's got all tubes. The other one is solid state and has more power for the hip-hop and rap and club stuff. The three key engineers here all use that same room to cut, and almost every day there is somebody cutting something. We were very, very surprised at how much is still going on in vinyl. I don't even know where you buy them [records] anymore, but I know they must be around somewhere.

There's one store down on Melrose [in Hollywood] that only has records.
BERNIE GRUNDMAN: Well, that might be where they are. But if the labels really merchandised them, they could probably sell even more because a lot of kids really like those things.

Most of the stuff we're doing is really high-end audiophile stuff on the tube system done from the original masters from the late '50s and early '60s, or we're doing almost like promo records, where they've got a 12" single with three or four cuts on there. We're doing more and more current albums too, and they don't even want to take tunes off to make them fit. On long CDs, we're doing them on four sides, and they're putting it on a gate-fold jacket. It's amazing; if an artist has any notoriety at all, they'll do it on vinyl as well as CD.

How do you think that having experience cutting vinyl has helped you in the CD age?
BERNIE GRUNDMAN: Well, the problem with vinyl is that it has more limitations than with CDs so it takes a lot more knowledge to cut a good vinyl disc than it does to do a CD. With CDs, except for artifacts and various changes that occur in the digital domain, what you get on the monitors is very close to what you get on the disc, and you don't have all the various distortions that vinyl can come up with. Vinyl has inner groove distortion and it has tracking distortion because of too much energy in the high frequencies. But this doesn't happen on CDs. With CDs, of course, the quality is the same from the beginning to the end of a CD, which isn't the case on vinyl. High frequencies might get a little brittle, but they don't distort on a CD, whereas they will on vinyl. So there is this whole grab bag of problems with vinyl that you have to consider. So part of being a good vinyl-cutting guy is knowing how to compromise the least.

All of us here have been in the business awhile and are very experienced with vinyl, so we can probably get about as much as you can out of it. But they're harder and harder to cut with the way these digital tapes sound. They have all of this energy now because people don't have to worry about being conservative on the bottom or the top end of a CD. Whereas if you listen to old vinyl discs, you notice that they don't have anywhere near the bass or high end that CDs have nowadays because there

was a cutting limitation. You just couldn't play a record back that had too much energy in the high end. That's why things have gotten so bright and aggressive on CDs I think, because now you can get away with it.

Talk about the level wars for a minute.
BERNIE GRUNDMAN: That's one of the unfortunate things about the industry, and it was even that way with vinyl. Everybody was always trying to get the loudest disk, and then if you got into a new generation of playback cartridges that could track cleaner, they would push it again until even those were on the edge of distortion. So it didn't matter if you had better and better cartridges because that just meant that you could go that much louder and get right up to the same amount of distortion you were at before. Hopefully it was louder than your competitors' records because that's a very basic, almost naive, kind of competitive area that people can identify.

Usually anything that sounds louder gets at least some attention. It might not hold up on the long haul, but the main thing that a lot of promotion guys want is to at least attract attention so that it gets a chance. What happens is everybody is right at that ceiling level as high as you can go, so now guys without a lot of experience try to make things loud and the stuff starts to sound god-awful. It's all smashed and smeared and distorted and pumping. You can hear some pretty bad CDs out there.

Would you have any words of advice for somebody who's trying to master something themselves to keep them out of trouble?
BERNIE GRUNDMAN: Well, I just don't think that you should do anything that draws attention to itself. Like if you're going to use a compressor or limiter on the bus, if you use it to the point where you really hear a change in the sound, you're going a little too far. You always have the consolation of knowing that the mastering engineer can take it to another level anyway, and he's experienced in how to do that.

Some of the automatic settings in these things really aren't as good as they make them out to be. And when you use them, you have to realize that you're going to degrade the sound, because compressors and limiters will do that. It's just another process that you're going through no matter if it is in the digital domain or analog.

This is another thing that is very true that I've studied for quite a while. Analog and digital are very, very much alike when it comes to signal processing. If you put an equalizer in the circuit, even if it's all in the digital domain, you will hear a difference. If you put a compressor in the circuit, not even compressing, you will hear a difference, and it will sound worse.

Do you do all of your processing in the analog domain then?
BERNIE GRUNDMAN: No, we do some processing in digital. We do compression and limiting sometimes in the digital domain because some of that stuff is pretty good if you use it right. But, our equalization is all analog because I have yet to find a digital equalizer that is as good.

What are you using for a compressor?
BERNIE GRUNDMAN: It's something that we have actually put together. It's kind of an oddball thing, but it works for us.

So you build digital gear as well.
BERNIE GRUNDMAN: Yes. We can hybrid stuff if we want. We could do part of the processing and even do the equalization in the digital domain if we felt we had a good equalizer. Our boards are built to accommodate anything you want because at some point we convert it to digital, and after that we can hang more stuff on it.

So the consoles are digital?
BERNIE GRUNDMAN: No, the main console isn't, but we have outboard equipment that we can put in the digital chain if we want. We have a whole desk area for digital stuff right next to the analog console so we could add in digital compression, limiting, or equalization if we wanted to.

How important is mono to you? Do you listen in mono often?
BERNIE GRUNDMAN: No, I very rarely listen in mono. Sometimes I do it just to test the phase, but I never listen in mono anymore.

What is the hardest thing that you have to do?
BERNIE GRUNDMAN: One thing that is really hard is when the recording isn't uniform. What I mean by uniform is that all of the elements don't have a similar character in the frequency spectrum. In others words, if a whole bunch of elements are dull and then just a couple of elements are bright, then it's not uniform. And that's the hardest thing to EQ because sometimes you'll have just one element, like a hi-hat, that's nice and bright and crisp and clean, and everything else is muffled. That is a terrible situation because it's very hard to do anything with the rest of the recording without affecting the hi-hat. You find yourself dipping and boosting and trying to simulate air and openness and clarity and all the things that high end can give you, and so you have to start modifying the bottom a lot. You do the best you can in that situation, but it's usually a pretty big compromise.

If the client just had a bright monitor system and everything in the mix was just a little bit dull, that is easy. It's almost like a tone control because you bring the high end up, and everything comes up. But when you have inconsistencies in the mix like that, it's tough.

Then there's something that's been overly processed digitally, where it gets so hard and brittle that you can't do much with it because once you've lost the quality, you can't get it back. If I am starting out with something that is really slammed and distorted and grainy and smeary, I can maybe make it a little better, but the fact that a lot of that quality is already gone is going to handicap that recording. It is never going to be as present as the way something that is really clean can be.

That is part of what gives you presence—when it's clean. The cleaner it is, the more it almost sounds like it is in front of the speakers because it's got good transients. Where if it has very poor transients, it just stays in the speakers. It sounds like it's just coming out of those little holes. It doesn't ever fill up the space between the speakers.

Do you have to add effects much these days?
BERNIE GRUNDMAN: No. Sometimes if it's lacking spatially really badly, we can put the B.A.S.E. [spatial processor] unit in. We have a couple of those around, and every now and then they come in handy because they can give a little more of an expansion to the ambience. But other than that, we don't. We almost never add echo either, unless it's like a classical recording where there are one or two instruments. There you can do it, but usually it messes things up if you try to put it on something that is really complex. It just confuses it.

What makes a great mastering engineer as opposed to someone who is just competent?
BERNIE GRUNDMAN: I think it would be what I was talking about at the beginning. I think it would be trying to get a certain kind of intimacy with the music. It doesn't even have to be music that you like. Music is a human expression, and you have to be willing to open yourself up to wherever it is that the artist is trying to go with their music or whatever he's trying to communicate. There is no reason why you can't get on that same wavelength, because you're also a human being, and we're all basically alike. But that is sometimes hard to do because you're not always on so you can't always do it. It's like any artist. They are not always on, and they're not always open to where their internal, basic humanity comes out. And that's the thing that will communicate to everyone because that's the thing we have in common.

So the real test is if you can really not be a snob, or not have all kinds of preconceived ideas, and just open yourself up to it and see how the song is affecting you emotionally and try to enhance that. I think that a lot of it is this willingness to enter into another person's world and get to know it and actually help that person express what he is trying to express, only better. I think that is a big factor when it comes to mastering.

You're going beyond the technical, in other words. You're going to the spiritual.
BERNIE GRUNDMAN: Oh, yes. Because that's what music is. My wife is an artist—she's a painter—and she has the same experience. When she goes in her studio, it's almost like it's not her painting when she's really on. And anyone that's played a musical instrument knows that there are these moments when it almost feels like you're not doing it. You're in touch with something really greater than you. It's going through you. It's a very elusive thing and hard to know how to get there. This is part of being concerned about how things are affecting others rather than just being all wrapped up in yourself.

How long do you think it takes to get to that point?
BERNIE GRUNDMAN: I think it varies. It depends on the emotional issues that people have—their personal defenses and their sense of self-esteem. Some people have such low self-esteem that it's really hard for them to even admit that there's a better way to do something. If a client suggests something, they're very defensive because they feel that they have to have the answers. A lot of engineers are that way, but mastering is more than just knowing how to manipulate the sound to get it to where somebody wants it to go.

We have a double board here where we can compare EQs, and one artist used to sit over there and do an EQ himself. I would do one on my side and then we would compare them to see who wins. Now a lot of engineers would be deathly afraid to do that because that would mean that, "God, what if he wins? That means I'm no good." That's low self-esteem. You think that if one thing is off or there's something that somebody had thought of that you didn't think of, that means you're no good. But maybe it's just how you're feeling that day. There are a lot of other things that you've done that are great. People have to know that about themselves. That one little thing that might not be right doesn't mean your whole world is gone.

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CHAPTER 21

Interview: Bob Katz

Co-owner of Orlando-based Digital Domain, Bob Katz specializes in mastering audiophile recordings of acoustic music, from folk music to classical. The former technical director of the widely acclaimed Chesky Records, Bob's recordings have received disc of the month recognition in *Stereophile* and other magazines numerous times, and his recording of *Portraits of Cuba* by Paquito D'Rivera won the 1997 Grammy for Best Latin-Jazz Recording. Bob's mastering clients include major labels EMI, WEA-Latina, BMG, and Sony Classical, as well as numerous independent labels.

What's your approach to mastering?
BOB KATZ: I started very differently from many recording engineers that I know. Number one, I was an audiophile, and number two, I did a lot of recording direct to two-track. That's my orientation. I am a very naturalistic person. I work well with rock & roll and heavy metal, but the sound and tonal balance of a naturally recorded vocal or naturally recorded instrument is always where my head turns back to. I find that my clients, while they don't necessarily recognize naturalistic reproduction as much as I do, love it when I finally EQ a project and make it sound what I think to be more natural.

Now, there are exceptions. A rock & roll group that wants to have a really big heavy bass, well, I'll go for that. But, at the same time, I'm more inclined toward projects that sound good when the EQ is natural.

Do you think there's a difference in the way people master from geographic area to geographic area? Do people master differently from New York to Nashville to LA, for instance?
BOB KATZ: Well, there used to be a West Coast sound.

Do you think there is now?
BOB KATZ: I think that I can identify the product of Doug Sax and Bernie Grundman a lot. But if you compare a lot of Ludwig against Doug Sax or Bernie Grundman, I think you'll find more similarities than differences, even though they're on different coasts.

I think that as the years have gone on, without mentioning names, some mastering engineers have succumbed more to the "crush it" campaign, while others are still holding their ground, and when that happens you hear a big distinction between engineers. But I see that same phenomenon on the West Coast as on the East Coast, as well as elsewhere. I think it's more of an individual mastering engineer in the fact that some of them happen to be located in the same location, rather than a city-by-city thing.

What do you think makes a great mastering engineer? What differentiates somebody who's great from somebody who's merely competent?
BOB KATZ: Great attention to detail and extreme persnickety-ness, stick-to-it-iveness, and discipline. The desire to just keep working at it until it's as good as the sound that you have in your mind, and to keep trying different things if you're not satisfied. I will bend over backward to get something right, even if I have to do it off the clock. Not to say that I don't charge for my time, but if I make a mistake or I feel that I could've done it better, the client will always get my best results.

As good as you have in your mind. Does that mean that before you start a project, you have an idea where you're going with it?
BOB KATZ: I think that another thing that distinguishes a good mastering engineer from an okay mastering engineer is that the more experienced you are, the more you have an idea of how far you can take something when you hear it and pretty much where you'd like to go with it, as opposed to experimenting with 10 different pieces of gear until it seems to sound good to you. That distinguishes a great mastering engineer from an okay mastering engineer in the sense that you'll work more efficiently that way. That's not to say that there aren't surprises. We're always surprised to find that, "Gee, this sounds better than I thought it would," or, "Gee, that box that I didn't think would work proved to be pretty good." And sometimes we will often experiment and say, "Let's see what that box does." So it's a combination of not being so close-minded that you won't try new things, but having enough experience to know that this set of tools that you have at your command will probably be good tools to do the job before even trying it. Also, a real good sense of pitch and where the frequencies of music are allows you to zero in on frequency-based problems much faster than if you have a tin ear.

It's hard to be in this business if you have a tin ear....
BOB KATZ: True, but I know a lot of medium-level people who get away without that degree of precision. There is another area, and that is the ability to be a chameleon and get along incredibly well with all different kinds of people from all walks of life. If someone brings in a type of music toward which I'm not necessarily inclined, I'll psych myself up and do pretty well with it, but I think that there are other people out there who perhaps do that even better than I do. So, being a chameleon and being adaptable and versatile is what distinguishes a great mastering engineer from an okay one.

What's the hardest thing that you have to do?
BOB KATZ: Make a silk purse out of a sow's ear. It's a lot easier to take something that comes in at an A-minus and turn it into an A-plus than it is to take something that comes in as a B-minus and turn that into an A. That is the hardest thing I have to do.

The next hardest thing is to teach my clients that less is more. When they're preparing their work to send to me, and also when I'm working on it, we'll often go in a big circle. I may know in my head that putting three different compressors in a row isn't going to make it better, but when they suggest it, I'll never refuse their suggestions. When it's all done, though, they usually realize that passing it through less is more. The exception being that Phil Spector kind of approach where you think that more is more, but in that case the purity of the sound is less important than the bigness and the fuzziness and all the other things that it does. That's not necessarily my kind of sound anyway. I'd rather make something sound really good and clean than good and dirty if I can.

What kind of project do you enjoy the most?
BOB KATZ: Music that is acoustic based. That doesn't mean that they don't have electric instruments, but there are musicians playing together, and the music's been performed all at the same time with few overdubs. I love those kinds of projects because I can really make them shine. Fortunately, people seek me out for that stuff, so I tend to attract that. It keeps me off the charts, though—darn it.

What makes your job easier?
BOB KATZ: This is almost becoming a ubiquitous answer, but I have to say that if I get the highest resolution, highest sample rate, earliest generation, uncut, unedited by anyone—or if they do cut it, leave the heads and tails alone—version, then things are easier. Unfortunately, I get more and more chopped up material these days.

For instance, I did a children's record, and Meryl Streep did the voiceover in a number of places. Now, they left her dry so if I needed to add reverb to put in between sections, I could do pretty much anything I wanted. But there were three cuts where they mixed the voiceover with the music, and when I finally put the CD in, three of the four worked fine in context with the songs they came in front of and after. But on the fourth one, the original mix engineer chose to mix the music fairly low against the voice and, after she finished talking, brought the music up to a certain level. When it was put in context in the mastering against the song before and the song after, the music was too low but the voice sounded at the right level when placed at the proper level to fit to the cut before.

I was stuck with a problem of the music being too low. So in my first revision I sent to them, I cheated the music up gradually after Meryl stops speaking, but not enough, because the cheat doesn't sound as good as if I had gotten separate elements and had been able to cheat the music up underneath without raising the voice.

So, what am I leading to is that you run into certain situations that are special or different. The problem is that many mix engineers don't know what *is* special or different. It's good to consult with the mastering engineer ahead of time, and in this case I would have said, "Send me the elements. Don't mix it, because when you finally put an album together in context is when you'll discover that you may need the separate elements." I think that the future of mastering increasingly will involve some mixing.

So you'd be getting stems essentially.
BOB KATZ: More often, and as we move to surround, we're going to be getting stems. I think that even two-track mastering will start moving into stems if we can ever standardize on a multitrack format.

If you get program material in that has already been edited (and of course a lot of times what they do is chop the fades), does that mean that you have to use outboard effects sometimes in order to help that along? And if so, how often do you have to do that?
BOB KATZ: More often than I'd like to. But sometimes the fixes are so good that the guys never realize how much they screwed it up when they brought it to me. I've always been a great editor, and that always helps. If you're good at editing, you can supply artificial decays at the end of songs with a little reverb and a careful crossfade that's indistinguishable from real life.

At the head of things, it's not as easy. The biggest problem with the headfades is that people just cut it off. The breath at the beginning of a vocal is sometimes very important. I think part of it is that number one,

they don't have the experience with actual editing over the years and don't recognize it as being an important part of the engineer's art. And number two, if you have a system such as Sonic or SADiE, you have great flexibility with crossfades. You realize that you can do things that other people can't, which is to carefully massage a breath at the beginning of a piece so that it sounds natural. But if you cut something—and not just the breath but something which I guess we would call the air around the instruments prior to the downbeat—it doesn't sound natural.

And how to fix that? Well, I'm not sure I can give a general answer. It's a lot easier to talk about how to fix fade-outs and end fades than it is to fix beginnings. The bottom line is, send us the loose material. If a client has a real good idea on the fade-out that they want to do, fine. Then send us both versions—the faded and the nonfaded. That way, if it proves to be a problem in context, we can still use the unfaded version.

What piece of gear are you using to help the fade-outs?
BOB KATZ: Being a naturalistic engineer over the years, the first digital reverb that I really felt sounded natural was the EMT 250 and its variations. Anyway, they got smaller and smaller and finally made a 32-bit unit that is only two U high that had the same sounds in it [EMT 252]. That was the first digital reverb that I felt sounded very natural, but I couldn't afford it at the time. So I was always searching for a poor man's EMT and renting them whenever I needed one.

A reverb chamber is used surprisingly a lot in mastering to help unify the sound between things. I might use it on five percent of all my jobs. So, I still needed a pretty good unit. Then I discovered the Sony V77, which is obsolete. After you spend a couple of hours fine-tuning it, it can sound just like an EMT.

I've heard that from other people as well.
BOB KATZ: It is really good. Now we're not talking about things that immediately attract people to a Lexicon, like smoothness and lack of flutter echo. Those are basic things that anybody can put into a reverb. What distinguishes the EMT and the V77 from the rest of the pack is the ability to simulate a space and depth. I've gotten it down so quickly that I can supply tails with a combination of Sonic and a few keystrokes in the verb, and it's all patched in in a matter of a minute or less for any tail.

What is your signal path like? Do you have an analog and a digital signal path?
BOB KATZ: Yes, but I'm a purist and I try to avoid doing an additional conversion whenever possible. The logical place to do analog EQ is when an analog source comes in. My analog path starts with a custom-built set of Ampex MR70 Electronics, which in my opinion are the best playback

electronics that Ampex ever invented. They were designed to be mastering EQs and there were only a thousand built. It has four bands of EQ itself—a high shelf, a high peak and dip, a low shelf, and a low peak and dip for the playback at 15 or 30 ips. I have that connected to a Studer C37 classic 1964 vintage transport with the extended low-frequency heads that John French put in, made by Flux Magnetics. It's just real transparent and not tube-y sounding at all, just open and clean. And nothing ever goes through a patchbay. It's all custom patched.

Usually I try to avoid any analog compression at that stage, and I try to make the tape sound as great as possible with either its own EQ or through the Millennia Media [NSEQ-2], so it's just real transparent. That goes directly, with a pair of short Mogami cables, into my A-to-D converter. So that's my analog chain. I don't have any other analog processing. I built a compressor once, but after playing around with the Waves Renaissance compressor and a few other digital compressors, I'm convinced that I'm just as happy staying in the digital domain once I'm already there. So at that point I convert with the best analog EQ possible, and the rest of the processing is done digitally after it's in Sonic.

Is most of your processing done prior to the workstation?
BOB KATZ: I think that there are two different types of engineers. I'd like to think the old-fashioned and the new-fashioned, but that's my slant on it. There are the engineers who like to process during load in, and there are the engineers who like to process on load out. Many engineers will set up an entire chain, either analog or digital or a hybrid of both, and process on load in, and then if it doesn't work in context, they'll go back and reprocess and then load it in again.

I find that to be a very inefficient way of working, so I'm really puzzled why they put themselves through this. The most I will do with the analog tape, as I said, is go through this great EQ on load in only because I don't want to go through another conversion again. After that, I favor having as many processors automated as possible. It just shocks me that there aren't that many mastering engineers who work that way.

I think that as the years go on, more and more mastering engineers will be working my way. I think they'll have to. When you start getting into surround, I think it's just going to become the norm. It's very much like the way you work with an automated mixing console.

How important is mono to you?
BOB KATZ: I forget to listen in mono more often than I intend to. I have good enough ears to detect when something is out of phase; it just sounds weird in the middle. In fact, I'm usually the first person walking into a stereo demo saying, "Hey, your speakers are out of phase." So I usually

don't have that much of a problem with mono, but I'm always using a phase correlation meter and an oscilloscope to make sure things are cool. If I see something that looks funny, then I'll switch to mono. But, half the time I just look at the scope and listen and won't switch these days.

Do you ever normalize?
BOB KATZ: "Normalize" is very dangerous term. I think it should be destroyed as a word because it's so ambiguous. If you mean do I ever use the Sonic normalize functions so that all the tracks get set to the highest peak level, the answer is no—I never do that. Do I use my ears and adjust the levels from track to track so that they fit from one to the other, then use compressors and limiters and expanders and equalizers and other devices to make sure that the highest peak on the album hits 0 dB FS? Yes, I do. I don't call that normalizing, though.

Tell me why you don't do it.
BOB KATZ: I'll give you two reasons. I advise my clients not to do it, and I've written about it extensively on my website [digido.com]. The first one has to do with just good old-fashioned signal deterioration. Every DSP operation costs something in terms of sound quality. It gets grainier, colder, narrower, and harsher. Adding a generation of normalization is just taking it down one generation.

The second reason is that normalization doesn't accomplish anything. The ear responds to average level and not peak levels, and there is no machine that can read peak levels and judge when something is equally loud.

Tell me how you came about choosing your monitors. And then, how would you suggest someone else go about it?
BOB KATZ: Let's start with the first question, which is a lot easier to answer. A great monitor in a bad room does absolutely nothing for you, so if you don't start with a terrific room and a plan for how it will integrate with the monitors, you can forget about it. No matter what you do, they will still suck and you will still have problems, so let's just say that I first started out by designing a great room.

The first test that anyone should do for a system is called the LEDR test. It stands for *Listening Environment Diagnostic Recording* and was invented by Doug Jones of Northeastern University. Basically, he determined the frequency response of the ear from different angles and heights. Then he simulated the frequency response of a cabasa if it's over your head, to your left, behind you, beside you, in the middle, and also beyond the speakers. In other words, from at least a foot to the left of the left speaker, over to at least a foot to the right of the right speaker, all done with comb filtering that simulates the response of what the ears would hear.

The LEDR test is a substitute for about $30,000 to $40,000 worth of test equipment. If the sound for the up image doesn't go straight up from your loudspeaker, six feet in the air as you sit there in your position, then you've got a problem with your crossover or with reflections above the loudspeaker. If the sound doesn't travel from left to right evenly and smoothly with the left-to-right test, then you've got problems with objects between your loudspeakers. And the same with the beyond signal, which is supposed to go from about one foot to the left of the left speaker, gradually over to one foot to the right of the right speaker, which detects reflections from the side wall.

So the first thing you should ever do as an engineer is to familiarize yourself with the LEDR test, which is available on Chesky Test CD, JD-37, and also on the ProSonus Test CD, which is about fifty dollars more. Just test your speakers and room with the LEDR test. And believe me, if you ever want to know how bad it can sound, just take a pair of cheap bookshelf loudspeakers and play the LEDR test through it and see what happens. It also shows how bad the lateral image is if you take a pair of monitors and put them on their sides with the tweeter and the woofer to the left and right of each other, as opposed to vertically.

So my room passes the LEDR test impeccably, so then it comes to the choice of loudspeakers. The speakers I chose are made in Switzerland by a man named Daniel Dehay. They're called Reference 3As (www.reference3a.com) and they are your classic two-way high-quality audiophile loudspeakers. I'm sure that there are about half a dozen high-quality audiophile equivalents from other manufacturers that can do just as well, but the whole thing is that these do not have a crossover per se; the woofer is directly connected to a pair of terminals in back of the speaker, and the tweeter goes through a simple RC crossover. They're wired to my Hafler amplifier. The woofer is an 8" speaker and it's ported in the back, and the speaker has a really tight, clean response down to about 50 Hz.

With an 8"?
BOB KATZ: Yeah, the guy did a really nice job. It's really an excellent speaker, the Reference 3As. But like I say, you can find some things that are reasonably equivalent. Right now, if somebody would ask me for a recommendation, I'd say PMC or the Dynaudio, and so on. Anyway, these speakers play loudly and cleanly without a problem since they have a 93-dB sensitivity. To top it off then, I have a pair of Genesis Servo subwoofers, and they have their own crossover amplifier. There is no separate high pass or bass management type of device on these speakers. I let the main speakers roll off with their own natural roll off, and then I carefully adjust the subs to meet seamlessly with them. I could go on, but I think that covers it.

You're running stereo subwoofers.
BOB KATZ: Right. That's absolutely essential.

What are you using for a console?
BOB KATZ: Aha! Mostly, you mean, for EQing and leveling and stuff?

Are you using a console at all?
BOB KATZ: No. I've never been impressed with the whole console concept. Most of the time I take the signal through the DAW desk at 24 bits with it set for unity gain so that it doesn't do any calculations.

The first thing that it feeds, nine times out of ten, is the Z-Systems equalizer. Then I patch various forms of external outboard digital gear using the Z-Systems digital patchbay and eventually bring it right back into DAW and cut the CD master.

How do you adjust the control room level?
BOB KATZ: I have an audiophile Counterpoint D-to-A converter with Ultra Analog Module, and it sounds as good as the Mark Levinson or one of those similar-quality D-to-As. I went into the Counterpoint and installed a stepped attenuator with metal film resistors at an interstage point. That is my volume control. It's calibrated in 1-dB steps, and the output of the DAC feeds my power amp directly. It is the cleanest, purest signal path that you've ever heard. So I have no preamp or no console, and I'm using absolute minimalist circuitry.

Well, I think the whole console concept is really a throwback to the lacquer days anyway.
BOB KATZ: Yeah, where you need a preview and all that stuff. Well, as we get into surround, we're going to need some console features. Mastering engineers are getting away from the console concept, although people like Bernie [Grundman] and Dave Collins will build a purist high-quality console because they want to do analog processing. I'll simulate that by patching gear one into the other into the other with short cable.

There are definitely two schools of thought on this.…
BOB KATZ: Yeah, they are real purists. But it just reminded me of something. I've been in many mastering studios, and almost every mastering engineer that I know of sits in front of some kind of a table, which sits at some height, with maybe a monitor in front of him. And then six or eight or nine feet in front of him are his stereo loudspeakers. As far as I'm concerned, there is some compromise there. Now anything that breaks into the listening triangle between my ears and my monitors is verboten in my studio.

My solution is that I have a listening couch where I and/or my clients sit, which is exactly like a high-quality audiophile living-room listening environment. We have the perfect 60-degree triangle there, with nothing in between except the floor and the side walls, which are far away from interference from the monitors. It's a reflection-free zone. Then, behind the couch is the back of the display of my workstation. When I want to edit or do some preliminary setup or segues, I go back there and do my primary work. It keeps my heart working. I get up, walk to the couch, sit down, listen, and go back. I don't EQ from back there, though, which prevents me from making those awful immediate judgments that are so often problems. Too many highs? Well, listen for a few minutes. "Oh, wait a minute. That was just the big climax with the cymbal crash."

I have a Mac PowerBook sitting on the arm of the couch connected by Ethernet to the rest of the system. I can remote control the Z-Systems equalizer from the arm of the couch, start and stop Sonic, or switch the Sonic desk between its record and playback desks, which allows me to monitor two different digital paths. So I can effectively insert or remove any set of equipment from my chain at the critical listening point without having any interfering tables or consoles in the way. Just a pair of function keys on the PowerBook, over there sitting on my right. Can you picture it? You're sitting there on the couch, your right arm is off to your right, and you just push a little button on a little portable computer sitting on the arm of the couch. And that's it.

CHAPTER 22

Interview: Bob Ludwig

After having worked on literally hundreds of platinum and gold records and having mastered projects that have been nominated for scores of Grammys, Bob Ludwig certainly stands among the giants in the mastering business. After leaving New York City to open his own Gateway Mastering in Portland, Maine, in 1993, Bob has proved that you can still be in the center of the media without being in a media center.

What do you think is the difference between someone who's just merely competent and someone who's really great as a mastering engineer?
BOB LUDWIG: I always say that the secret of being a great mastering engineer is being able to hear a raw tape, and then, in your mind, hear what it could sound like, and then know what knobs to move to make it sound that way.

You know where you're going right from the beginning then, right?
BOB LUDWIG: Pretty much. It's a little bit like the Bob Clearmountain school, where after 45 minutes of mixing he's practically there and then spends most of the rest of the day just fine-tuning that last 10 percent. I think I can get 90 percent of the way there sometimes in a couple of minutes, and just keep hanging with it and keep fine-tuning it from there. It comes very, very fast to me when I hear something. I immediately can tell what I think it should sound like. And the frustration is, sometimes you get what I call a "pristine piece of crap." I call it that because it's like a bad mix, and anything you do to it will make it worse in some other way. But 99.9 percent of the time, I hear something and I can figure out what it needs, and fortunately I know what all my gear does well enough to make it happen.

Like today, I was doing something while training one of the guys that works with me. I put this song up and said, "I know this piece of gear would be perfect for this thing." He said, "Man, I haven't seen you use that in like nine months or a year." I said, "I know, it's gonna be great." I fired it up, plugged it in, and boom, it was right there.

How many of your sessions are attended?
BOB LUDWIG: When I started my own business after working at Masterdisk and Sterling Sound before that, our business plan called for a 20-percent reduction in overall business, but the opposite actually happened. We thought that half the people who had attended sessions in New York would attend up here. It turns out more people attend sessions here than in New York, which was a total surprise.

Why do you think that is?
BOB LUDWIG: I'm not sure. To tell you the truth, I think a lot of people have heard about the effort we've gone through to make our room as acoustically perfect as possible. And they know that we've got speakers that retail for $100,000 a pair, so a lot of people just want to come and see what it's about.

It's a real pleasure. So many times people come into the room and they go, "Oh, my God!" or something like that. It's a trip to get that kind of reaction from people. When I was at Sterling and at Masterdisk, everybody thought I owned those companies, but I never did, and to me it was always frustrating that I was always dependant on my employers dictating my conditions. That was one of the reasons I left. I felt that if I stayed in New York, I'd never be able to have a room that was acoustically as perfect as we knew how to make it. I don't know about the new place, but Sterling and Masterdisk always were in highrises, so you're always limited to very low-ceiling rooms. But in order to get as near-perfect a situation as possible, you actually need a fairly large shell that's at least 30 feet long and accommodates a 17- or 18-foot-high ceiling.

Do you think that there's a difference between the ways people master from coast to coast?
BOB LUDWIG: I don't think there's so much a difference between coast to coast as there is just between some of the major personalities in mastering. Some engineers might master almost everything into the analog domain because they love working with analog gear. I certainly do that sometimes, but I would say that I've tried to accumulate what I think is the very best new gear as well as funky old gear that has a certain sound. If a tape comes in sounding really, really good, I have gear that will stay out of the way and do exactly what I need without inflicting any damage on the thing at all.

Occasionally we'll get a tape in that's so good that I'm just happy to change the level on it if needed. The level controls I have are made by Massenburg and some engineers over at Sony and are as audiophile as you can get. If you're not using the level control, you can take it out of the circuitry so it's as much a straight wire as possible, so at least I'm convinced I'm inflicting as little damage as possible on a great-sounding tape if all it needs is simply a level change.

Is that in the digital or the analog domain?
BOB LUDWIG: Analog. Talking about different engineers, there are some engineers who just like to slam the hell out of everything. It seems like their only criterion is how loud they can make it, not how musical they can make it. And for me, I'm under pressure from A&R people and clients to have things loud, but I try to keep the music at all costs. I'll think nothing of doing a Foo Fighters record one day, where it's totally appropriate to have it smashed, then the next day do something that's perhaps even 4-dB quieter than that because it suddenly needs the dynamics for it to breathe.

The dynamics wars… where did that come from?
BOB LUDWIG: I think it came from the invention of digital domain compressors. When digital first came out, people knew that every time the light went into the overs or into the red that you were clipping, and that hasn't changed.

We're all afraid of the over levels, so people started inventing these digital domain compressors where you could just start cranking the level up. Because it was in the digital domain, you could look ahead in the circuit and have a theoretical zero attack time or even have a negative attack time if you wanted to. It was able to do things that you couldn't do with any piece of analog gear, including an Aphex Compellor or [Empirical Labs] Distresser. It will give you that kind of an apparent level increase without audibly destroying the music, up to a point. And of course, once they achieved that, then people started pushing it as far as it would go. I would say the average level of a CD has peaks on a VU meter that are at least 3.5 dB hotter than they used to be, if not as much as 6 dB hotter than they used to be.

I always tell people, "Thank God these things weren't invented when the Beatles were around, because for sure they would've put it on their music and would've destroyed its longevity." I'm totally convinced that over-compression destroys the longevity of a piece. Now when someone's insisting on hot levels where it's not really appropriate, I find I can barely make it through the mastering session.

Another thing that has contributed to it is the fact that in Nashville, the top 200 country stations get serviced with records from the record company, but apparently there's some kind of an agreement that the major record companies have for stations 201 on up to get serviced with a special CD every week that has the different labels' new singles on it.

It's called CDX. Glenn Meadows does that.
BOB LUDWIG: And of course, when they started doing that, the A&R people would go, "Well, how come my record isn't as loud as this guy's record?" And so that further led to level wars even in Nashville, so that

everyone's record would be the hottest record on the compilation. And of course when the program director of the radio station is going through a stack of CDs, a mediocre song that's twice as loud as a great song might at first seem more impressive, just because it grabs you by the neck. It has a certain impressiveness about it, so you listen to it before realizing there's no song there, but at least on first listen it might get the program director's attention.

I suppose that's well and good when it's a single for radio, but when you give that treatment to an entire album's worth of material, it's just exhausting. It's a very unnatural situation. Never in the history of mankind has man listened to such compressed music as we listen to now.

In mixing too, if you don't put buss compressors on, or if you don't compress something, clients inevitably say, "Why are you not doing that? That's what I want." You can't get into trouble if you squash something, but you can if you don't.
BOB LUDWIG: I know some very famous mixers who complain to me about A&R people who will not accept their mixes unless they already sound as though they have been mastered, already devoid of any dynamic range.

Do you think we've reached the limit of that?
BOB LUDWIG: Yeah, I honestly do, because we're not that far away from music dynamics approaching steady-state tone! If you look at many of today's CDs on a digital level meter, the peak levels barely go lower than the maximum. It would be a steady stream of digital "over"-levels if the digital domain compressors didn't artificially prevent the red "over" light from coming on. It's difficult to believe that it could be compressed much more than it is now. That's why I'm so excited about 5.1, because there's no radio competition.

You mentioned about people asking you to add reverb and effects. Does that happen often?
BOB LUDWIG: Oh yeah, it happens often enough. Speaking of Pro Tools, a lot of people assemble mixes on Pro Tools, and they don't listen to it carefully enough when they're compiling their mix, and they actually cut off the tails of their own mixes. You can't believe how often that happens. So a lot of times we'll use a little 480L to just fade out their chopped-off endings and extend naturally. I do a fair amount of classical music mastering, and very often a little bit of reverb is needed on those projects. Sometimes if there's an edit that for some reason just won't work, you can smear it with a bit of echo at the right point and get past it. Sometimes mixes come in that are just dry as a bone, and a small amount of judicious reverb can really help that out. We definitely need it often enough that we've got a 480L in our place, and it gets used probably once every week.

Do you still get in projects mixed to both analog and digital?
BOB LUDWIG: Yes, but at 88.2/96, it's often a tossup. Sometimes the digital sounds better; sometimes the analog sounds better. A lot of it depends on who the mixer is. Some of the premier mixers, like Bob Clearmountain, get exactly what they want on digital tape. He sends me stuff at 88.2/24-bit, and I'm sure it's a very, very close match to what comes out of his console.

For most engineers, analog tape serves as wonderful kind of acoustic glue that sounds better than the output of the console. Analog is very forgiving, and our ears really seem to love it. We place a lot of attention on analog at our place. We've got six different ways of playing back analog tape. We've got a stock Studer A820. We've got a Studer that's got Cello class-A audiophile electronics. We've got a stock ATR, a tube ATR, and an unbalanced ATR. We also have one of the Tim de Paravicini 1" two-track machines with his fantastic tube electronics. When you record with his custom EQ curve at 15 ips, it's basically flat from eight cycles up to 28 kHz. It's unbelievable. You put an MRL test tape on his machine, and it comes back 0 VU all the way.

Tell me about your monitors.
BOB LUDWIG: I used to have Duntech Sovereign 2001 monitors. I think around '86 when I was at Masterdisk, I decided to find the best monitors I could so that when I was working on digital I would have something that could really reproduce subsonic defects. So I went down to New York to some of the audiophile shops to see what kind of audiophile speakers I might be able to find for mastering that would be professional enough that I wouldn't have to change the tweeter every other day.

I found these Duntech Sovereign 2001 speakers. Tom Jung, the engineer that owns the DMP label, had a pair at his house in the basement. His basement had very low ceilings. The Duntech speakers are in a mirror-image arrangement; the tweeter is in the middle, and then there are the midrange speakers, and then there are the woofers on the top of the speaker and the bottom. So in the basement of his house, that upper woofer was coupling with his ceiling, as well as the bottom one coupling with the floor, and he had bass for days. So he sold me his pair of Duntechs, and that's what I used at Masterdisk from then on.

I also bought one of the first Cello "Performance Amplifiers" from Mark Levinson when he was there at the time, and subsequently he told me that somebody in Japan had actually bridged a pair of these things, and it was really worthwhile. Of course his amps are mega-expensive, so he loaned me another pair so I could try to bridge them together. Doug Levine, who ran Masterdisk and was in charge of all the money, could

actually hear the difference between the bridging and the non-bridging enough that he thought it was worth spending the extra money on it.

Then when I started Gateway, I got another pair of Duntech Sovereigns and a new pair of Cello Performance Mark II amplifiers this time. These are the amps that will put out like 6,000-watt peaks. One never listens that loudly, but when you listen, it sounds as though there's an unlimited source of power attached to the speakers. You're never straining the amp, ever. So I used those Duntechs for quite awhile.

Then, when I began doing 5.1 surround music, Peter McGrath, a classical engineer friend of mine, had fallen in love with these Eggelston Works Andras speakers that are made in Memphis. Bill Eggelston has been designing speakers for many years, and Peter told me that he thought those were the best speakers that he had heard at the time. Peter used to own an audiophile hi-fi shop, and he's heard everything under the sun. As he's a very good classical engineer, I give what he says a lot of credence. So I had made it a point to seek them out. I really fell in love with these Andras, and for the 5.1 music, I use five of them. They retail for around $14,000 a pair, and I have 2-1/2 pairs of them. They were *Stereophile* magazine's speaker of the year. With five of them in the room, they move plenty of air with no problem whatsoever, but I felt that there needed to be a bigger speaker to work right in stereo.

I told Bill Eggelston if he ever decided to build a bigger version of the Andras to let me know, and maybe I'd consider changing my Duntechs if I thought they sounded better. He decided to build what he thought was the ultimate speaker, which is called the Eggelston Works Ivy speaker. (He names all of his speakers after former wives or girlfriends.) These speakers are a little bit taller than Duntechs and they weigh close to 800 pounds a piece. They've got granite on the sides of them. There are three woofers on the bottom, a couple of mids, the tweeter, and then a couple of more mids on the top. Actually, each cabinet has 23 speakers in it.

You know how M&K uses the isobaric principle in their subwoofer? The Eggleston Works Andras use that same isobaric principle in their woofers. Well, Bill extended that principle to all of the speakers, so behind each speaker are two others. I guess if the isobaric principle is carried out to purity, you'd have an infinite number of speakers. But he has two behind each of them, and they're amazing. Every client that comes in, once they tune in to what they're listening to, starts commenting on how they're hearing things in their mixes that they had never heard before, even sometimes after working weeks on them. It's great for mastering because they're just so accurate that there's never much doubt as to what's really on the tape.

One reason I've always tried to get the very best speaker I can is I've found that when something sounds really right on an accurate speaker, it tends to sound right on a wide variety of speakers. I've never been a big fan of trying to get things to sound right only on an NS-10Ms.

Do you listen only with that one set of monitors or do you listen to near-fields?
BOB LUDWIG: Primarily just the big ones because they tell you everything, but I do have a set of NS-10Ms and some ProAcs and stuff like that. Lower-resolution near-fields have their place. In the case of the NS-10Ms, the reason we have them there is just so the client can hear what he *thought* he had on tape! The NS-10M kind of dials in a little bit more reverb than you think you have and more punch than is really there. When I'm teaching people, I make sure that they listen on NS-10s and ProAcs and speakers like that a lot, so they can learn in their head how to translate from one to the other.

Do you think that having experience cutting lacquers helps you now in the digital domain?
BOB LUDWIG: It does. I'm certainly more concerned about compatibility issues than a lot of the mixers are, especially as more people are getting into either QSound or other kinds of synthetic ways of generating outside-of-the-speaker sound. Some people just get into this and don't realize that their piano solo is gone in mono. It just happened to me recently. A very famous artist came in, and the piano solo had this wild spatial effect on it, and the piano was just not there when you listened in mono, so I had to point it out to them. And much to my surprise, they said, "We don't really care." Well, people do still listen in mono, but some artists just don't seem to be bothered by the lack of compatibility. Nevertheless, I'm probably more hypersensitive to sibilance problems than I would otherwise be if I hadn't cut a lot of disks.

Does that mean you still listen in mono a lot?
BOB LUDWIG: I certainly check in mono. We have correlation meters on our consoles. Even though my room is huge, QSound works perfectly in it on the large speakers because the first reflections are so well controlled. So any time there are QSound-like effects, one can hear it in a jiffy. In my room, if you're sitting in the sweet spot and flip the phase on one of the speakers, the entire bass goes away. It's almost as if you were doing it electronically. So you can hear any phase problems instantly, and then of course you just monitor in mono. Plus, I have the ability to monitor L minus R as well to hear the difference channel if I need to.

Tell me about your signal path.
BOB LUDWIG: In the analog domain, it goes from the tape machine into George Massenburg/Sony electronics that are as minimal and audiophile

as one can get. The output of that goes into a dCS, a Pacific Microsonics, or sometimes an Apogee analog-to-digital converter. When I need other outboard gear, we've got Neumann EQs and NTP and Manley compressors. Between the Manley, NTP, and digital domain compressors, that normally fills the bill for me, but I do have some Aphex Compellors. In the digital domain I have all the Weiss 96/24 stuff. The bw102, which has the 96-kHz de-esser in it as well, is complete with a mixer, compressor, and equalization. We use a lot of the Waves products because they are 48-bit internally and sound good.

Do you have a Waves L2?
BOB LUDWIG: Yeah, we have three of the production units and one of the beta versions right now. We also have SPL units, and before that we had the Junger units.

What's the hardest thing that you have to do? Is there a certain type of music or project that's particularly difficult?
BOB LUDWIG: I think the most difficult thing is when the artist is going through the period where they just can't let go of the project. You get into the psychological thing where in the same sentence they say, "I want you to make the voice more predominant, but make sure it doesn't stick out." Just contradictory things like that. They'll say, "This mix is too bright," and then you'll dull it up like half a dB, and they'll say, "Oh, it doesn't have any air anymore." It's that kind of thing.

Letting go is so hard for some artists. One of my favorite artists is Bruce Springsteen. I think he realizes mastering means he has to finally let go of the record and crystallize it. I think, unlike new artists, he has the ability to put out the record exactly as he wants to, and I've seen him live with records for a long time as a result. And in his case, he's correct in not putting it out until he is completely happy.

Do you have a specific approach to mastering?
BOB LUDWIG: To me music is a very sacred thing. I believe that music has the power to heal people. And of course a lot of the music that I work on, even some of the heavy-metal stuff, is healing some 13-year-old kid's angst and making him feel better, no matter what his parents might think about it. So I treat music very, very seriously.

I love all kinds of music. I master everything from pop and some jazz to classical and even avant-garde. I used to be principle trumpet player in the Utica, New York, Symphony Orchestra, so I always put myself in the artist's shoes and ask myself, "What if this were my record? What would I do with it?" So I try to get some input from the artist. If they're not there, at least I try to get them on the phone and just talk about what things they like. I just take it all very seriously.

CHAPTER 23

Interview: Glenn Meadows

Glenn Meadows is a two-time Grammy winner, a multi TEC award nominee, and former owner of the famous Nashville-based Masterfonics Studios. He has worked on scores of gold and platinum records for a diverse array of artists, including Shania Twain, LeAnn Rimes, Randy Travis, Delbert McClinton, and Widespread Panic, as well as for producers and engineers, such as Tony Brown, Jimmy Bowen, and Mutt Lange.

What's your philosophy on mastering?
GLENN MEADOWS: I think that mastering is, and always has been, the real bridge between the pro audio industry and the hi-fi industry. We're the ones that have to take this stuff that sounds hopefully good or great on a big professional monitor system and make sure it also translates well to the home systems. We're the last link to get it right or the last chance to really screw it up and make it bad. And I think we're all guilty at times of doing both.

That being said, do you listen on typical home hi-fi systems?
GLENN MEADOWS: No, my old mastering room at Masterfonics had an in-wall Kinoshita monitoring system. It's about an $80,000 or $90,000 speaker system when you include the amplification. What we found is that when you have it sounding really great on that, it sounds good on everything else you play it on. Yeah, it's a different characteristic than a home system without the dome tweeters and that thin, ethereal top end that comes out of there, but if the components in the big system are in good shape and they've been maintained properly, you're going to get that same perspective. It also doesn't rip my head back and forth trying to go to different monitoring systems.

What I think is really difficult is that if you put up two or three different monitors to get a cross section, then you don't really know when anything is right because they all sound so different. I used to run little B&W

100s and I'd also have the requisite NS-10s in the room, and during that time when I was switching back and forth, I found my mastering suffered radically because I didn't have an anchor anymore. I didn't have a point where I knew what was right because the character of the speakers was so different from each other. Once you listened to one for a couple of minutes, you lost your reference point on the others.

The reason people come to a mastering engineer is to gain that mastering engineer's anchor into what they hear and how they hear it and the ability to get that stuff sounding right to the outside world. So if you start putting all this stuff up on small speakers and try this and try that, you've basically created a big confused image for the mastering engineer.

Well, that being said, does that mean you only listen on one pair of speakers?
GLENN MEADOWS: Yeah.

So you never go to a smaller pair?
GLENN MEADOWS: I do at home. I do in the car. I do outside of the mastering room. I'll pop it on in another room in the building. All of our rooms are cross-connected fiber-optically so we can literally walk into another room and dial the first room up and listen on those speakers. It's really very handy having that. But in the room itself when I'm working? No, it's the one set of monitors.

If I get a producer that says, "Well, I've gotta listen on… fill in the blank," then we get a pair, and it's like, "Okay, here's the button that turns them on. Here's how you start. Here's how you put the EQ in and out if you want to listen that way. Call me when you're finished listening." And I leave the room and let them listen because it literally rips me away from my anchor. If I start listening on different-sounding monitors, then I'm completely lost. But on the monitors that I've worked on for 13 years in the same room, I know how they sound. I know what they need to sound like, and the repeat clients go, "Yep, that sounds right. Yep, that sounds good." What you find is typically within a song or two of working with somebody who has been in here, they settle into it and say, "Okay, yeah. I really can hear all that detail. I understand exactly what you are doing." We put other things up for them to listen to that they're familiar with to get a cross-check on what I'm used to hearing.

Do you think that there's a difference in the ways people master from town to town? Is there a difference because of where you're geographically located?
GLENN MEADOWS: I don't think that's as much true anymore as it used to be. I could probably put a vinyl record on and tell you where it was mastered and who did it. To some extent the early CD transfers were very similar to that as well.

Right now, though, it's all blended in to be a big jumble of sound, and you almost can't pinpoint anybody's characteristic fingerprint anymore. Everybody has basically the same kind of tools and is doing the same kind of thing to satisfy the customers. And, unfortunately, satisfying the customers is, in my opinion, not where the music needs to be right now, but that is a whole other story.

Let's go there. What brought that about, do you think?
GLENN MEADOWS: The level wars? We had level wars in vinyl right near the end of it, where everybody was trying to get the vinyl hotter and hotter and hotter. And at least in vinyl you had this situation where when the record skipped, the record label would say, "Well, it's too loud and you're gonna have returns." What put the fear of God into the producer was returns. By God, we don't want any returns. So they would tend to back away, and we could kind of stay within the limits of the medium, where you got a 23-minute side here and you couldn't cut it any hotter because it just won't fit at that level. Those were the realities that you had to live with.

We originally thought we had that type of limitation on digital, but what ended up happening is there's so many tools out now for doing the dynamic range squash that you can literally get tracks now where you put them in a workstation and it looks like a 2×4. It comes on at the quietest passage at the beginning of the intro and it's full level. You get into what I call "dynamics inversion." Spots in the record that should get louder actually get softer because they're hitting the compressor/limiter too hard.

I don't think that the record companies and the producers at this point have enough insight or understanding about what radio learned a long time ago, which is the tune-out factor for distortion. Radio has spent a lot of time researching how far you can push it before people are annoyed and won't listen anymore. As a result radio is tending to back down a lot with their compression, but it still gets compressed when they mix it, we compress it when we master it, and they compress it when they broadcast it. If you look at some of the radio stations on a VU meter on a calibrated system, they have maybe 3 dB of dynamic range.

I'm mastering one right now that's a French-Canadian album, and it's a joy to listen to because it's got dynamics. It's an independent release by an artist from Canada. It's great; it has dynamics. It lives. I challenge any mastering engineer to go back and listen to music that they did four or five years ago, when they were putting greatest-hits packages together, and listen to the mastered versions compared to what they're getting now. Then ask themselves, "Have we really gone forward or have we've gone backward?" This happens to me all the time.

Whose fault is it?
GLENN MEADOWS: I think it's a wrap-around effect from broadcast. To be very honest with you, there is the impression that if the song doesn't jump off the CD for the program director's initial listen, then he's going to hit the "next track" button. So, we get into this round-robin deal where we've got to make the cuts louder and louder so that they jump off the CDs faster.

We used to do an every-other-week compilation called CDX, which is a collection of all of the country stuff coming out in the next four- or five-week period and is a service to all the non-reporting Billboard R&R type stations. The labels actually buy slots on this so it relieves them from having to send release CDs on singles to thousands of radio stations. All they then have to concentrate on are the 150 or 200 reporting stations because this service handles the 2,500 others. So they buy a slot on this for every one of their releases. We compile it for them, and we have ever since it started. The sequence of the songs on the CD is alphabetically based by song title, so Aaron Tippin doesn't always go first, or Arista Records doesn't always get their stuff first. Every single release is a jumble, so there's no preferential positioning on the disc. We've spaced those five, six, eight seconds apart, trying to make them less like an album so it's just like a collection of songs.

All the producers and the record labels get copies of this, and the first thing they do is compare their cut to somebody else's, and if theirs isn't as loud, they go back to their mastering engineers and say, "What's wrong with this?" Or they call us and say, "You screwed mine up. You didn't make mine as loud." Wait a minute—all we're doing is compiling. If you do a digital compare, what's on the CD is exactly what was given to us by whoever mastered it. We don't play with it, we don't change levels, and we don't have preferences. We are a fulfillment center, and that's all we're doing, so don't blame us. So they go back to their mastering engineer and say, "The next time a track is going on CDX, make sure it's good and hot." So we get specialized releases for CDX that have been run through additional processing and have even less dynamic range. Then you have the situation where the record label listens to this advance copy and pulls out their mastered album and says, "The one on the CDX is louder than the one on the full album. Why is that?"

Catch-22.
GLENN MEADOWS: It's a complete catch-22. I just had a 1" two-track rolled in while we're talking because I'm re-mastering a ref on a shootout again. They came to me first, and everybody loved it. One of the people involved in the project said, "Well, I really think we ought to go over here [to a competitor] to master." So they pulled the tapes, went over to this other place, took my ref, and said, "Here's what he's done. Can you beat

it?" So of course, he got more level and they said, "Wow, look at that." So the producer and the head of the label said, "You know, we really like what you did, but we don't feel it's fair that you went first. Do you want to take another shot at it and hear what the other guys did?" So here we go. The tapes are coming back today. I'm going to get a copy of what they did to see if I think I can do it any better or any differently. But the irony is that the producer was here when we did it the first time. This is what he said he wanted. Now, why are we doing this again? The problem is if you stay in a situation where you're always going first and end up not doing the mastering, then you have people go, "Well, why should I even go over there?" It's a horrible situation, and I don't personally know how to break the cycle, other than getting people to listen.

As the quality of the music is going down, so are the record sales. I don't think anybody has tried to make a correlation between the fact that if it's fatiguing to listen to, the people at home are going, "I can't even listen to the whole record. It comes on, it's in my face—it never gets quiet, there are no dynamics. I could only listen to five songs. Take it off and throw it away. It's irritating."

Do you think the problem lies in mastering or is it in mixing?
GLENN MEADOWS: God, that's a hard question.

I must admit that if I don't use the buss compressor, I have clients who will get upset. And no matter how bad it sounds, you never get in trouble if you use it. But you get in trouble if you don't.
GLENN MEADOWS: Right. And of course you alter your mix because it's in there, so it wouldn't do any good, really, to have one without it because it's not going to have the right balances. It really is a catch-22.

My typical approach to do that is to use like a 1.15:1 compression ratio and stick it down at –20 or –25 so you get into the compressor real early and don't notice it going from linear to compressed and basically just pack it a little bit tighter over that range. I'll get maybe 3 dB of compression, but I've brought the average level up 3 or 4 dB, and it just makes it bigger and fatter. That's what we did to it, and the record label goes, "Wow, how did you do that? It doesn't sound limited and compressed?" And he and I just looked at each other and smiled. It sounded great on the radio, and that's the whole point. People think that they have to be heavily compressed to sound loud on the radio, and they don't.

When you use your compression technique, are you using the typical radio attack and release settings? Long attack, long release?
GLENN MEADOWS: No, it varies. It depends on what the tempo of the music is doing. I'll adjust it track by track.

Breathing to the music.
GLENN MEADOWS: Yes. Most everything I do is tailored to what the music dictates that it needs. There's no preset standard that I'm aware of that I use, although I had a producer come in and have me master a record, and then he went back and matched it with a Finalizer and stored the setting: "Ah, there's the Masterfonics setting." He told me he did the same thing for Gateway. He had a couple of things mastered up there and then found a common setting, and now he's got it as his Gateway preset. He does his own mastering now. "Ah, make it sound like Gateway. There it is." I told Bob [Ludwig] that, because he and I have been friends for probably 20 years, and he just died laughing. He said, "If you can find out what that setting is, send it to me. I'd love to have it, because I don't know what I do."

What makes a great mastering engineer?
GLENN MEADOWS: The ability to use discretion. The ability to listen to a piece of product and say, "You know, this really doesn't need much of anything." At this point in my career—I've been doing this for almost 30 years now—if I put a client's tape up and I don't have a pretty good clue by the time I'm at the end of the first run of the first song as to what that song needs, they ought to go back and remix. I find that the real value of a mainstream mastering facility versus trying to do it yourself or doing it in a small backwoods-type place or a basement place is that the experience of the engineer comes into play, and it can save you money and time. We have had situations where clients say, "Oh, we can't pay your $210 an hour. We know how long it takes to master." And I said, "Well, tell me about what you did the last time." "Oh, we went to this guy and it was $25 an hour." "How long did you spend?" He said, "We spent four days." "Three or four hours a day?" "No, he worked 10, 12 hours a day. It cost us a fortune." I'm just shaking my head in disbelief and saying there is no reason that an album of what you're putting out should take more than seven or eight hours at the most. I said, "To be real honest with you, if I had to spend more than four or five hours on the record to get 98 percent of what can be gotten out of it, I'm wasting your time."

I don't mean to be arrogant, but it has to do with the experience of the engineer working in his environment. He's in the same room every day for years. I can walk into this room in the morning and know if my monitors are right or wrong just by listening to a track from yesterday. To me, that's the value of a mastering engineer. What they bring to the table is the cross section of their experience and their ability to say, "No, you really don't want to do that."

Speaking of which, what makes a great facility? Is it possible to have a great mastering engineer in a facility not up to par with his abilities?
GLENN MEADOWS: Yes, it can be done because he knows the facility and he knows its limitations—how to work around them and how to get the

most out of the facility. You can put a mediocre engineer in a great facility, and if he doesn't know what he's doing and doesn't know how to get the most out of what tools he has at his disposal, you are never going to get there.

Tell me about a great facility. What makes it great?
GLENN MEADOWS: It is not something that necessarily has the latest and greatest bells and whistles. It's a facility that's able to capture what you started on tape and see it through to where the client is happy with what he walks out the door with, and the ability to do that on a consistent basis as well. It doesn't necessarily have to be exactly right the first time because that's why you give a client a reference. You let them go listen in the environment they're familiar with, because you're forcing them into your environment to start with. That's why they've come to you, because they value your opinion and your ears and what that brings to the table. By the same token, we all can't expect to get them 100-percent right every single time.

What's your typical day like?
GLENN MEADOWS: For me, usually in by 8, 8:15. I get caught up on last-minute projects where clients might need some copies by mid-morning or there's an emergency single that gets pulled from an album—that type of stuff. If it's a day with clients, then we pretty much try to hold to one project a day unless they are singles, then maybe we do two or three a day. If it's an album, we'll start at 10:00, break at 12:30 or 1:00 for lunch, then come back and finish up. In the afternoon we're running references for the client. If we're done early, then we're able to get onto our production work for albums that are approved or references that need level tweaks or changes done to them. We're kind of unique in Nashville in that we're very close to Memphis so our FedEx pickup is 9:15 at night, and it allows us to run a long day. If we finish with clients at 4:00, we can then start cranking out whatever it may be that has to go out the door. Whereas on the two coasts, the last pickup is at 5 PM, so if you miss that, it's like another day before it gets done. Here, we've got another four-hour shift that we can run.

Do you do your own production work?
GLENN MEADOWS: I do my own production work. That's just part of what I bring to the table with the clients. I've got a lot of hard drive space available, so a lot of projects can stay online for a long time until they are approved. When we're doing CD-R masters, we run them at real time with audio present so that we can hear what goes down, because I'm the one who did the work so I'm the one who is going to notice if something is wrong. If I pass this onto somebody, and there's a process that's not working right or an automation move that sounds weird, they're not going to know. After all, it's my name that goes on the project as "mastered by." I

did the same thing on vinyl for quite a few years when I did all of my own lacquer cutting.

What do you usually send to the replicator?
GLENN MEADOWS: Verified masters are run through the StageTech verifier, the printout sheet is put in a Ziploc bag, and the jewel box is taped closed with a note to the plant saying that if it's opened and there's a problem, we don't warranty it. We tell the client, "If you take it out and play it, it's yours. If there's a problem on it, we've verified it. We've listened to it by ear." While we verify we also have a guy listening on headphones for any extraneous clicks or pops or anything strange, so it's been listened to twice. That's why we charge $350 for it. We run into those situations where a client will say, "Oh, just give me a ref disc," and they'll take the ref and approve it and send it straight on to the plant. It leaves you kind of like scratching your head, going, "Okay, but how do you know if the disc is good?" because we don't run the CD-R references through the verifier.

You should catch something by that time.
GLENN MEADOWS: Absolutely. And he also checks that the start cues are working, and then we look at the printout of the error report and make sure there are no extraneous E22s and things like that on the disc that should reject it. If we catch that, we just burn it again. But we don't want the client to listen to it because they've already approved a reference disc, and they're paying us to make sure that their master is what they've approved. That's the value we bring to the table, rather than cutting the CD master and saying, "Okay, here it is. You go listen to it and decide if it's okay." That, to me, is passing the buck.

We're getting paid a large number of dollars to do these. They look at us like, "$350? The disc only costs two bucks." And I say, "Yeah, but you're not paying for the disc. You're paying for the time it takes us to create it to give you exactly what you are supposed to have." So that's kind of the way it works, and we don't have any problem with clients trying to listen to them as a result. It gets to the plant, and the plant says, "Yes, it came in sealed," so it seems to be working.

Is there a particular situation that's more difficult than others?
GLENN MEADOWS: Probably I put myself in the situation where I continue to work with custom people—guys who are just putting out 500 CDs. I've always felt that they deserve as much of an opportunity to have their product handled by a pro as anybody else does. But you get some stuff and you just kind of have to roll your eyes like, "Wow, this is really bad." You have to be diplomatic about it because that's the best a client can do sometimes. I think that's the hardest thing; being diplomatic in situations when you know that in reality they are only going to sell these to their friends and family.

Do you do a lot of these?
GLENN MEADOWS: I do enough of them. I used to not be available to do that type of stuff, and I personally felt bad because part of how I started out in this business was doing custom disc mastering. These people want to pay the rate, so they deserve to have what can be done to help their product. In many cases it's a whole lot easier to make dramatic improvements on bad-sounding stuff than it is to take something that sounds great and make it dramatically better. That's even harder—to try and make a dramatic improvement in a great-sounding tape and to know when to leave it alone.

What do you enjoy the most in mastering?
GLENN MEADOWS: I enjoy anything that is well recorded and the music is good. Be it a French Canadian project I'm listening to while we do this, be it a jazz thing or a classical project. If the music is good, I really enjoy it. We do most of the mastering on the Cirque de Soleil soundtrack albums for their shows, and that is just a joy to work on because the music is great. There is no pretense that we're trying to make this radio-friendly or anything else. This is a piece of music that has got to sound great at home, and that is the enjoyable part, when it doesn't have to be commercial.

Is there something that a producer can do beforehand that makes your job easier or something that just makes it a lot harder? Maybe that's two questions.
GLENN MEADOWS: I really hate, and have a much more difficult time, working with material that has been pre-pre-mastered. I'm not crazy about any of those mastering-in-a-box type deals, because most of what they do is undoable. Most people using them are listening in less than ideal environments, and they can't hear a lot of the stuff that's going on. Plus, your ears become so used to it that it becomes like an addiction where more is better. If it is louder, it is better. If it has got more bass and more top, it is better. Just whatever more is, that is better. As a result you have a file that is sitting right at zero or clipping, and they want you to master it. You're going, "Well, there's barely much left to do. You have kinda killed it already."

Do you normalize? Do you ever use the normalize function?
GLENN MEADOWS: No. I don't use a computer to decide how much to bring something up. Typically, I will process on the way into the workstation. I am not a load-it-in-and-then-master kind of guy. I prefer to take the original source material and go through whatever processing gear I decide I need or would like to use on the project, and come into the workstation and deal with it that way.

I have a reference point where I park the monitors when I start working on the cut, and I kind of get a feel for what it is doing and then look at the headroom coming in to see where I am at. Invariably, I end up within 1/4 or 1/2 dB at the top, maybe because there is a little bit of a peak limiter sitting there as a protection. But once in the workstation, I will use the processing only as subtle final tweaks. I don't use the internals of the workstation as my mastering tools per se. The workstation is an editing area. It is a scratchpad to do all the work in and compile it and put it together. The outboard gear is what I use for mastering, and that is just the way I have grown into it.

Is that signal chain digital, analog, or a combination?
GLENN MEADOWS: It can be a combination, but my path is typically 99-percent digital because 99 percent of what I am getting is digital.

Do you ever get a request to add effects or have to add some tail to something that has been cut off?
GLENN MEADOWS: Every now and then we do, yes. We just did a thing where one of the cuts on the album is a live piano/vocal track done live at a show. The mix that they ended up doing was a bit too dry, so we just added some verb and mastering to it, and they are all happy.

Generally, what do you use?
GLENN MEADOWS: I use a Lexicon 300L if we need it and route through the mixer in SADiE. In this particular case, the stupid little plug-in that SADiE had gave just the character it needed, so it literally was added inside the workstation and is part of the project, which in itself is strange, but it works.

Do you use subwoofers?
GLENN MEADOWS: No. The monitors in the room I am in and the room measure flat to 28 Hz.

CHAPTER 24

Interview: Bob Olhsson

After cutting his first number-one record (Stevie Wonder's "Uptight") at age 18, Bob Olhsson worked on an amazing 80 top-10 records while working for Motown in Detroit. Now located in Nashville, Bob's unique view of the technology world and his insightful account of the history of the industry makes for a truly fascinating read.

How do you think mastering has changed from the vinyl days to the way it is now?
BOB OLHSSON: Well, I was thinking about that. In the vinyl days we were very concerned with mechanics, meaning the playability of a record and whether it could be manufactured. A mastering error in those areas would mean thousands of returned pressings. It was a big financial factor. Tapes, for the most part, came from larger studios with more experienced people, so you didn't really have that much to do in a lot of cases. You might use little EQ, a little level correction, filter some low-frequency and de-ess some highs so you wouldn't run into skipping problems, but other than that you pretty much tried to go with the sound on the master tape. It was a lot more nuts and bolts. You'd always think, "How do I get it off from the tape onto the disk and still have something resembling the same thing come back?" So it started out very much as that kind of consideration.

Then, as the recording industry moved to the use of independent studios, we began to get a new generation of independent mastering studios. They got more involved with working on the audio itself, partly because the studios either had less experience or had less feedback than, say, you would get in a record company studio. In a record company studio, you hear about it in a big hurry if something doesn't sound good, whereas in an independent studio you may or may not hear about it because by the time the salespeople are involved, the studio is completely out of the loop. So Sterling Sound and the Mastering Lab and so forth were kind of the first generation of mastering studios that were not part of record companies.

At the same time, the record company studios became more involved in what we called “creative mastering.” This was where Bernie Grundman at A&M, for example, made a very large impact from a record company studio. On the East Coast I guess Sterling was probably the first. There was a studio, Bell Sound, which was both a recording studio and a mastering facility, and they were a very big deal. Motown used to send their stuff to Bell.

In 1948, the majors decided they were going to stop doing anything other than middle-of-the-road pop music, and so a whole bunch of people left the majors and started the independent record companies—the Atlantics, the VJs, the Chesses, and so forth. Later on, Motown was actually part of the second generation of that evolution. This was a whole parallel thing that was created by the advent of tape recording. The idea that you didn’t have to record to disk and go through all that stuff that required this specialized expertise was a revelation. You could now go into a studio that had done broadcast advertising, or you could go into a radio station. Atlantic used to use radio stations all over the country. They would find an artist they wanted to record and sign them to a contract on the spot. Then they’d find a local radio station, make a tape, and send it back to New York. A lot of their early records were done that way. They eventually built their own studio, and the rest is history. A friend of mine, Joe Atkinson, was their mastering engineer from 1959 until he came to Motown around 1969.

When you were at Motown, were you in Detroit or LA?
BOB OLHSSON: I was in Detroit, the real one.

You did the mastering?
BOB OLHSSON: Well, it was a complicated thing. Basically, Berry Gordy is a man who tried to never make the same mistake twice, so he had his own system that was integrated into RCA’s manufacturing. If at all possible, he wanted the mixes to be able to be mastered flat. So in many cases, if it didn’t work well flat, it got sent back to mixing rather than attempting to fix it in mastering. He also had a policy that he wouldn’t evaluate anything other than off a disk since he wouldn’t have a tape recorder in his office. He wanted to hear how it stacked up against other records on the market, and he wanted that perspective on everything he listened to. So we basically did an acetate of every mix that was done. We would occasionally suggest a change, but for the most part they wouldn’t approve anything at all radical. Anything beyond a couple of dB at 4,000 was sent back for another mix.

So what I was doing was basically cutting these acetates. We would cut a 33 1/3 of all the mixes, and then they would pick which ones they wanted to go to the next step. If there was some marketing reason why it had to

happen fast, we would do the mastering. But if there was time, we would send the acetate and the master tape to RCA and tell them to match it. They were willing to absolutely guarantee pressings and turn around any mistakes in 24 hours. We went that route because Berry's first business was a record store, and he knew all about defective pressings.

What was the reason for them doing the mastering? Did he think that there would be fewer rejects if it happened there?
BOB OLHSSON: He had a guarantee. Basically, the way it was set up is we would hardly even know about a problem because they would deal with it all internally at RCA. So they were actually matching an acetate that we had sent, and we would check their acetate to make sure that it matched what we had done before letting it go. That was the process.

That's far different from what you would think.
BOB OLHSSON: Yes, it was pretty unique. Basically, the secret of the success of Motown was being able to coordinate appearances of the artists with records in the stores at the right time.

You saw firsthand something that may not ever happen again. That was probably a wonderful experience to live through.
BOB OLHSSON: Oh yes. I'm convinced Berry Gordy is absolutely the smartest person I've ever heard of in the record business. All my experience since then has been looking at how people are doing things and scratching my head and wondering why on earth they are taking the long way around. I've watched various labels go through their changes, and my perspective is sort of an odd cynicism because I haven't seen much new. I would love to see somebody put together a book about how he actually ran the company. They have done all these books that have been basically written for the fans of the artists, but they haven't really gotten into how the company worked and what they did.

The neat thing about doing mastering there was that we saw everything. We had to relate to virtually every part of the company, and we were among the only people that ever saw the whole thing. It was really brilliant. Of course, I am also not sure that he realizes how brilliant it was. He was just a very bright and very, very, very logical man. He was always thinking, "How can I make this simpler? How can I make this better?" And it meant that we did everything in a somewhat different way than the rest of the industry, but often it was a much smarter way.

Like, for example, the Motown artists never paid for any studio time. They never paid for promotion. They didn't pay a manager's fee out of the record royalties. They didn't pay for a lot of stuff, and they got a lower royalty rate as a result. But you have all these people running around believing they really got ripped off because they don't realize that the higher

rates that the other companies paid would then get whittled down to next to nothing. So, it's an apples and oranges thing.

I was doing mastering there until about 1968, and then I got moved into the studio because I had a background in music. So from that point on I was doing vocals, strings, horns, rhythm dates, the whole bit. I was one of the two people that held every engineering job there. The other one is Larry Miles.

The musicians were all jazz players. Berry is a big jazz fan. His record store was a jazz record store, and it completely failed, but he learned his lesson. Just because he loved something didn't mean that it was commercial, so after that he began doing the most universally commercial stuff he could. His goal for the company was for it to be another RCA or Columbia.

And he almost got there.
BOB OLHSSON: I think what finally brought it down was the whole MTV thing, spending hundreds of thousands of dollars on videos and that kind of thing. Of course Motown was much more oriented around the music than the video.

I think the one effect of the Internet may be to completely turn that back around again. I think in a lot of ways it is like '48 all over again—the numbers aren't going to work for these big new conglomerates, and a new complete independent scene will develop. I look at online to play the same role that radio did in the '50s.

The thing people don't understand is that music is a social thing. People do music with other people. They want to hang out with people that are into a given kind of music. It's something they have on in the background of their life. It's like a piece of architecture almost. It's not something where they put their life on hold to concentrate on it, like a film. It's a very, very different product, and Motown was really aware of that. That and the dancing.

In retrospect, another thing is blatantly obvious, but I don't think anybody really realized it back then. What we called the R&B chart was really the women's chart. [Laughs.] I think the thing we didn't realize was that beginning with the Beatles, men had become an important component in buying records, and the records we were making largely appealed to women. We weren't all that successful at making records that men were into. That just kind of came crashing home to me recently. It's like our own racism limited us because we thought it was a racial thing and it probably wasn't. That may be true of the whole industry. Now it's swung back that way again. This last year, women just started buying more records than men for the first time since the Beatles.

I read somewhere that the demographic that buys the most CDs nowadays is white women over 30.
BOB OLHSSON: It's the fastest-growing group, I know that. I've actually been trying to research that some myself. In our web mastering project, one of the things that I have been doing is trying to come up with statistics about signal processing and demographics. Unfortunately, most of the research has been done by broadcasters and is extremely proprietary. They paid for it and they're damned if they're going to have other people knowing what they learned.

I had an exchange with Bob Orban [whose Optimod compressor/limiter is at the heart of most radio and TV stations' signal chains] and found out a couple of real interesting things. Apparently too much high frequency absolutely kills you with women, but a lot of bass is very important to women. Too much compression kills you with women because it becomes what he calls "intrusive." You want it to be able to be on and in the background all the time. You don't want it pulling your attention away. You still don't want it to be boring, and dynamics actually help with that, so it's a fine balance from a station's viewpoint. In order to appeal to women, they have to be less in your face, and the more in-your-face thing has to do with maybe the first 10 seconds that somebody listens to a station before they adjust the volume control.

How do you think we're going to get back to the use of dynamics, because now we're squeezing the life out of everything everywhere along the line?
BOB OLHSSON: The usual theory is that nobody will question it as long as it is selling, but, of course, new recordings are not selling. I found out that the average new release is selling something like 800 copies. The few titles selling very well, the recordings that are selling millions of copies, are not paying for the ones that aren't. Apparently this came up in SoundScan, and *Billboard* printed the thing, and a bunch of the majors tried to actually get them to pull that issue off the stands because they didn't want their stockholders seeing that statistic. So there is certainly something going on there.

I have heard that there are some major meetings going on in an attempt to more or less reel production back into the record companies. They are rethinking a lot of stuff because of the dropping percentage of titles that are paying for themselves. It may all come out in the wash because while stuff certainly is going to get squeezed, if people can come up with figures that indicate that over-compression can harm sales, that is definitely the message that can turn it around.

Returns would scare people away from going too far.
BOB OLHSSON: You had that same economic with vinyl. But in this case, we can do things beyond anything we were ever able to do before, like turn

the signal into a square wave, even. The other thing is that people are commonly going too far with compression during mixing, so much that an awful lot of mixes can't be helped. I average a couple of mastering jobs a year where I can't do anything to it. If you switch anything in at all, it just absolutely turns to dust. All you can do is hope that the stations that play it won't destroy it too much more.

Do you have a philosophy about mastering?
BOB OLHSSON: Well, first, do no harm. To me it's a matter of trying to figure out what people were trying to do, and then do what they would do if they had the listening situation and experience that I have. I sort of try to be them because I see the whole process as a matter of trying to clear the technology out of the way between the artist and the audience. You've got this person on this end who is doing a performance, and you have these people on the other end who are listening to it, so I think it's largely about keeping the technical aspects from distracting from the performance. That's the most basic thing. Then, to a certain degree, you can enhance things, of course. You can get it so that you can hear more of what they were doing on a wider range of playback systems or playback circumstances.

What I'm doing is mostly turning parts up, turning parts down, putting different EQ on different parts, and trying to get the dynamics so that there are some. I'm really trying to make something that somebody got working on a pair of Genelecs work on big systems and little ones, but yet somebody at a listening station in a record store won't need to switch the volume control. So it has to be up at the current accepted level, and yet I have to try to figure out how to do the least harm to it and still have it be an experience that people want to hear repeatedly. I can't understand the idea of somebody buying something that they aren't going to want to listen to over and over. To me, that is kind of the whole point.

But the big thing is communication. It's about somebody working some magic in front of a microphone, and people having the effect of that magic coming out of a loudspeaker. To me, that is the key to the whole thing. Do everything you can to get the music to happen in front of the mics and everything you can to protect it after it is an electrical signal.

The whole thing is to try to maximize the amount of expertise that you can afford, because you don't really want to master your own recordings. For my own recordings, if I can push the budget, I go to Bob Ludwig. I'm frankly more impressed with his work than almost anybody I have heard, and I have taken projects to just about everybody in the business. I think the man deserves his reputation. The unfortunate part of it is that at this point I suspect he gets mostly save jobs. Stuff where you'll never know how bad it really was. And so a lot of the stuff that has his name on it is fairly

mediocre and often was probably sent to somebody else, and the label bounced it back and said, “Well, okay. Let’s throw the big bucks at this and see if he can save it.”

What makes a great mastering engineer as opposed to someone who is just competent?
BOB OLHSSON: A willingness to go the extra mile and really dig in and try and make something better. It’s a willingness to fix the intro of something that is a little off as opposed to just letting it go.

How long does your typical mastering job take?
BOB OLHSSON: For independent clients, typically at least six hours.

Do you have to add effects at all?
BOB OLHSSON: Like reverb? Yes, we do that on some things. We do a lot of compilations where we’re starting with wildly different sources and trying to get them to lay together. It can be pretty challenging. We just did a compilation of some Russian choral music where some new recordings had been done in a pretty dry church, and they just didn’t mix with the stuff that had been done in a cathedral, so I had to add a ton of reverb to that.

What did you use?
BOB OLHSSON: Well, we have a NuVerb sitting in a spare machine, and that appealed to me because you can save the settings. Of course in mastering, a whole lot of what it’s about is how you reproduce it five years later. So I’m very, very anal about archiving source files and settings, and even software in some cases, so that I can pull it back later. Because as things have progressed, I’ve found that I can go back and take something I mastered five years ago and do a heck of a lot better job today. So if I can go back to the sources and even just see what my settings were, I can just use newer software. The software that’s made most of this happen is the Waves stuff.

Are you using just one set of monitors, or do you go back and forth?
BOB OLHSSON: I don’t like multiple monitors in a studio, although I’ll use the little speaker on a Studer two-track. I also check things out in my car. I find mid-level alternate monitors just confuse things.

Do you listen in mono much?
BOB OLHSSON: Yes, because too many decisions are made in mono down the line. We have had occasional problems. We had one artist that decided they liked the effect of the lead vocal 180 degrees out of phase on each side, so when you mixed it to mono it went away. We had to explain to them that you don’t really want to know what the limiter at a radio station is going to do on that, because the stations have these correlation

switchers that try to switch everything in phase. I understand there are also things that will somewhat mono-ize a signal because it will reduce the distortion in stereo. So there is a lot of manipulation going on there. They assume a clean, coherent signal going in, so if you give them something that isn't, heaven only knows what will happen.

How do you see mastering changing in the future? What will the mastering facility of the future look like?
BOB OLHSSON: I think there is going to be a lot more involvement by the producers and mixers than there has been because if any of the new formats fly, things are going to be a lot more complex. Having three different mixes of voice up, voice down, and voice in the middle in a six-channel surround is going to be pretty unwieldy to keep straight. I mean, there are just so many more things that can go wrong that I think a lot of it is very likely to go the way of the film business, because that was how they worked out to deal with all the different theatrical formats. Film mixes are done to stems, and then those are "mastered" to the various surround formats.

What are you listening to at home?
BOB OLHSSON: Duntechs with a pair of Hafler 9505s. It's real good for digital because it's a very bright, clean system, so it really shows up any artifacts. That's basically what we want it to do. We just want to come up with digital stuff that doesn't bite.

CHAPTER 25

Interview: Doug Sax

If ever there was a title of “Godfather of Mastering,” Doug Sax has truly earned it, as evidenced by the extremely high regard in which the industry holds him. One of the first independent mastering engineers, Doug literally defined the art when he opened his world-famous The Mastering Lab in Hollywood in 1967. Since then, he has worked his magic with such diverse talents as the Who; Pink Floyd; the Rolling Stones; the Eagles; Diana Krall; Kenny Rogers; Barbra Streisand; Neil Diamond; Earth, Wind & Fire; Rod Stewart; Jackson Browne; and many, many more.

Do you have a philosophy about mastering?
DOUG SAX: Yes. If it needs nothing, don’t do anything. I think that you’re not doing a service adding something it doesn’t need. Mastering doesn’t create the product. I don’t make the stew; I season it. And if the stew needs no seasoning, then that’s what you have to do. If you add salt when it doesn’t need any, you’ve ruined it. I try to maintain what the engineer did. A lot of times they’re not really in the ballpark due to monitoring, so I EQ for clarity more than anything.

When you first run something down, can you hear the final product in your head?
DOUG SAX: Oh yes, virtually instantly. Because for the most part I’m working with music that I know what it’s supposed to sound like. But once in a while I’ll get an album that is so strange to me, because of either the music or what the engineer did, that I have no idea what it’s supposed to sound like, and I often will pass on it. I’ll say, “I just don’t hear this. Maybe you should go somewhere where they’re clued into what you’re doing.”

But for the most part, I’m fortunate to usually work on things that sound pretty good. I do Bill Schnee’s stuff and George Massenburg’s and Ed Cherney’s and Al Schmitt’s—who’s the most nominated engineer, you know. I’ve done his stuff since 1969. These are clients that I’m the one they

go to if they have a say in where it's mastered. Every room has its claim to fame, and mine is that I work on more albums nominated for engineering Grammys than any other room, and probably by a factor of three or four to the next closest room.

How has mastering changed over the years, from the time you started until the way it is now?
DOUG SAX: My answer is maybe different than everyone else's. It hasn't changed at all! In other words, what you're doing is finessing what some engineer and artist has created into its best possible form. If an engineer says, "I don't know what it is, but the vocal always seems to be a little cloudy," and I can go in there and keep his mix and make the vocal not sound cloudy, that's what I did in 1968 and that's what I still do. The process is the same; the goal is the same. I don't master differently for different formats. I don't master differently for CD than I would for an LP because you essentially make it sound as proper as you can, and then you transfer it to the final medium using the best equipment.

There's a three-CD set which is a lifetime retrospective of Linda Ronstadt. I had mastered, I would say, 95 percent of all the originals, starting from *Heart Like a Wheel* when she was on Capitol Records, because I've done most of Peter Asher's [Linda's producer's] work. So it gave me a chance to look at this stuff that I had done in the '70s. Most of these tapes have the original EQ notes in them. My equalizers are the same as they've been for 30 years, so I could put on the tape, line up the tones, and throw up what I had done in '75 or in '78 or in '81. I would make some changes if necessary, but for the most part, what felt good then feels good now.

What surprised me is I had done a lot of work on my analog machines since then, and some of the tapes sounded absolutely better than in 1975 or in 1983. I could play them better today, so I was quite surprised how good some of those tapes sounded.

Did that influence any of your decisions then, because the stuff was coming back cleaner and better?
DOUG SAX: No. I just got more enjoyment out of it. Maybe a couple of times I took a dB of top off because I felt like I was getting more off the tape than I did then. Or maybe I felt that it could use a dB more bottom than I had done in '75. I've read articles in all the trade magazines about how the mastering engineer had to roll off the bottom to fit it on the disk, and now that we have CD you don't have to do this. And I think, "Who are they talking about?" I never filtered my low end for an LP, and I cut a very wide stereo. So I was wondering who they were talking about when they said that, now that supposedly you can really hear the full bottom because it didn't have to get all rolled off to fit onto an LP record. I was shocked at that.

Do you think that working on vinyl has helped you in these days of CDs? Would that experience help a mastering engineer?
DOUG SAX: I don't know if working on vinyl helps. I think having worked on many different types of music over the years helps. In one sense, being from the vinyl days, I was used to doing all the moves in real time. I never went down a generation. In other words, a lot of mastering places would make a fade on a tape copy, then they would assemble a copy and cut from that. I never did that. I always cut directly from the master tapes, so if you blew a fade on the fourth cut, you started over again. So the concept of being able to do everything in real time instead of going into a computer probably affects the way I master because I don't look at things as, "Oh, I can put this in and fine-tune this and move this up and down." I look at it as to what I can do in real time.

I find the idea that you have a track for every instrument and you put them all together to have great clarity doesn't work. I think it works the opposite way. The more you separate it, the harder it is to put together and have clarity. So if you're EQing for musical clarity to hear what is down there, that's unchanged today from way back 30 years ago. It's the same process. And the EQ that would make it clear, that would make somebody call up and say, "Wow, I really like it. I can hear everything, and yet it's still full," is still as valid today as it was then.

I'll tell you what the biggest difference is today from back then. The biggest thing is dynamics. There is no dynamic range now, and nobody wants dynamic range.

Why do you think it has changed?
DOUG SAX: I think I know precisely why it has changed. It has to do with the fact that there's an increasing amount of music listening being done in the car, and there's one thing that doesn't work in the car and that's dynamics. Long, sexy fades that ease you out of one song and into another are worthless in the car.

The thing that brought this to mind was when I was working on a critical album for a pretty famous engineer. We had done a couple of changes, and he came back and we did a couple more changes. Finally, we got to the point that the last change was made, and he called up in about an hour and said, "I love it. Don't touch a thing. It's done." And I said, "How can you judge? You haven't even been home yet." He says, "Oh, I do all my listening in the car. In fact, my home stereo hasn't even been on in a year." I'm not going to mention his name, but he's a major engineer who wins Grammys for his engineering, so it really brought to mind that I do my own listening in the car. I get stuck in traffic, but for recreation I listen to music that I don't normally work on, which is symphonic music. That's my background. I was a symphonic trumpet player, and you know Bob

Ludwig is a trumpet player. And I think Ted Jensen is a trumpet player as well. I don't know what it is, but trumpet players seem to make pretty decent mastering engineers.

What's the hardest thing that you have to do?
DOUG SAX: I come from a time when an album had a concept to it. The producer worked with one engineer and one studio, the group recorded everything, and there was cohesiveness as to what was put before you. Once you got into where they were going and what they were doing, you sort of had the album done. The multiple-producer album to me is the biggest challenge because you might have three mixes from Nashville in different formats, a couple from New York, and two that are really dark and muddy, and three that are bright and thin. The only good part that I see about this is that you absolutely have to have a mastering engineer. There's no question, the mixes don't go together and they don't work. The hard part is to find some middle ground so that the guy that has the bright, thin tape is still happy with what he's done and doesn't drive off the road when the dull, thick one hits after the bright, thin one. So that is the biggest challenge in mastering, making what is really a cafeteria sound feel like a planned meal.

I'm very proud of the fact that I've trained a lot of good mastering engineers, and I'll tell them, "You're not going to learn how to master working on a Massenburg tape. It's pretty well done. If he didn't like it, he wouldn't have sent it. But you get engineers that are not great, or you get these multiple-engineer things, then you can sort of learn the art of mastering by making these things work using your ears." Otherwise, it's pretty easy.

Were you the first independent mastering engineer, or one of the first?
DOUG SAX: Absolutely. Independent has to be clarified because if you go back to the late '60s and before, everything was in house. You were signed to a label. You were given an A&R man. You stayed at the label. You recorded at Capitol. You went down to Capitol's mastering to get your product mastered to lacquer. You went to Capitol's art department, and they gave you an artist that designed your cover and that's the way it was. It was really at the end of the '60s that certain top producers would say, "I love the security, but I would like to work with an artist that's not on this label. I would like to work with Streisand, but she's on Columbia." So they started to break off and really started the process where nobody uses label stuff for anything anymore. "If you sign me, I'll use the engineer I want, and I'll record and master where I want." That's 30 years of hard-fought independence. So from the standpoint of an independent that is not aligned with a label, just a specialty room that handles mastering, the answer is yes.

I was one of the pioneers when there was no business. We opened up our doors on December 27th of 1967, and by ’71, ’72, you couldn’t get into the place. By ’72 we were doing 20 percent of the top 100 chart, and there weren’t a lot of competitors. There was Artisan in LA, and Sterling and maybe Masterdisk just starting in New York. That was it. Now there seems to be a thousand, because the reality is that it’s very easy for someone to go into this business now or do it themselves. You can get a workstation with all the bells and whistles for a song and a dance. A Neumann lathe setup in 1972 was $75,000, and that was just the cutting system. You still needed a room and a console. So there were only a few people doing it, and you had to have a big budget. Now you fire it right up.

And don’t forget that in the industry, for almost 10 years there were no tones on an analog tape, so you didn’t know how to line up to the machine.

There were no tones?
DOUG SAX: No tones. I’m one of the instigators in railing on these guys to go back and print the tones so I could at least get my machine to be where your machine was. And there was no such thing as near-field monitoring. It didn’t exist. So people used to go to these strange studios with big speakers in the wall, most of which were useless as far as relating to the world, and the engineers never knew that they were out in left field because they had nothing to take home. The cassette was just starting, and only handful of engineers that I can think of actually had a 15-ips [inches per second] tape machine at home that they could take home a mix and find out where they were.

I started the process in the early ’70s just in self-defense. I would say, “Look, before you do anything, come in on the house with your first mix and find out if you’re in trouble. We’ll listen to it and get you straight.” I just got tired of watching these guys’ eyes open the first time they ever heard it out of the studio. “Oh my God. I couldn’t hear any highs in the studio, so I kept adding highs. I asked the guy, ‘Are these monitors right?’ and he said yes.” That absolute horrendous reality is the reason, really, why near-fields came in.

The truth of the matter is that the tools are getting so much better. I hate to say this as a mastering engineer, but used right, the Finalizer can do some awesome things. There was nothing like that three years ago. Digital technology is moving so fast, and it has gone from, in my view, absolute garbage to, “Hey, this is pretty good.” They’re getting better clocks on the computers. They’re getting better signal processing and better DSP. What used to be something that was really unmusical to me, if I have to say it, is now getting there.

I look at the Finalizer. A lot of mastering engineers badmouth it, and I get a kick out of that because with the Finalizer, you can make your product loud instantly. Mastering engineers don't like that because they used to be the ones that made it loud. But the reality is that everyone's going to have it and, as a result, everyone can make their CD loud. Once that becomes absolutely no trick at all, then the question becomes, are there things that maybe we should do besides just make it loud? I'm hoping that there's still going to be a business for someone that treats the music with love and respect when they're mastering it. And I think there's going to be a small reversion away from, "I want the loudest CD."

I get people in here new off the street that say, "I want the loudest CD ever made," and I say, "You're in the wrong place." Once in a while, they'll pull out a CD and put it on, and it's absolutely blazing, and I'll say, "Find out where that was mastered and go there and get what you're looking for." But as I say, I still do more Grammy-nominated albums for engineering, so I have to be competitive from the standpoint that you don't want to turn it up a bunch when you put the thing in a CD player.

Your reputation is that you're more of an analog guy....
DOUG SAX: My partner and I did some of the pioneering work in digital in the late '70s. The classic 3M machines [digital tape machines] were designed out in Camarillo, and my partner lived in Camarillo and did the original piano tests for them in '78. The very first recordings that were done on the Soundstream machine [the first digital recorder], before it was even up to a 44.1k sampling rate, we participated in. It was done right down the street at a church here. So when I'm being critical of digital, it is because I really have heard digital from the beginning, and I knew that it was not up to the best of analog. But we're talking about 1980, and there's been a lot of development since.

I get a lot of 96/24 stuff in. It's cheap, it's here, it's now. So any comments that I make about a Sony 1610 from 1985 that was absolutely just horrible then are true. And when I say that a 96/24 recording done with dB Technology converters sounds terrific, that's also true.

Describe your signal chain, or is that proprietary?
DOUG SAX: No, it's not proprietary. As a point of interest, whether the source is analog or digital, if it needs EQ, I EQ it as an analog. That makes sense because if you come in with 96/24, I just look at it as good-sounding analog. I do what I want with it, then I'll get it down to 44.1 and 16 bit in the best way possible. So whether it's 1/2" or 1/4" analog or digital, it goes into good converters and comes up as analog. Then the EQ is passive with the same equalizer I've had since 1968. The limiters are all tubes, and they're transformer-less. Ninety-nine percent of what I do is done between those two devices.

What do you use for monitors?
DOUG SAX: I use my own. They're two 15's with a midrange horn and a tweeter, and they've been here since 1968. I have no near-fields.

That's fantastic that what you have has weathered the test of time.
DOUG SAX: Yes. It's the same concept that I have about mastering. I don't master any differently today than I did in 1968. The speakers allow me to put the right stuff on, and if they steer me wrong, then they're worthless.

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CHAPTER 26

Interview: Eddy Schreyer

Noted veteran engineer Eddy Schreyer opened Oasis Mastering in 1996 after mastering stints at Capitol, MCA, and Future Disc. With a list of chart-topping clients that span the various musical genres, by such artists as Babyface, Eric Clapton, Christina Aguilera, Fiona Apple, Hootie and the Blowfish, Offspring, Korn, Dave Hollister, Pennywise, and Xzibit, Eddy's work is heard and respected worldwide.

Do you have a philosophy about mastering?
EDDY SCHREYER: Yes, I do. I would say the philosophy is to create a sonic product that gives the song balance and competes with the current market in terms of sonic quality and level.

What do you mean by balance?
EDDY SCHREYER: Frequency balance—not too much bottom, not too many mids, and not too much top. Balance is making adjustments with compression, EQ, and such so that it maintains the integrity of the mix, yet achieves balance in the highs, mids, and low frequencies. I go for a balance that is pleasing in any playback medium that the program may be heard in. And obviously I try to make the program as loud as I can. That still always applies.

But all mixes can't be cut as loud as others, so there are many limiting factors as to how loud something can go, and there are also limiting factors on what balance can be achieved. Some mixes just cannot be forced at the mastering stage because of certain ingredients in a mix. If something is a little bottom-light, you may not be able to get the bottom to where you would really like it. You have to leave it alone so it remains thinner because it distorts too easily.

There are a lot of people who are complaining that things are so squashed these days and it's because of everyone trying to get their competitive level up.
EDDY SCHREYER: What I am hearing is that various houses are really over-compressing trying to get more apparent level. The tradeoff with excessive compression to me is the blurring of not only the stereo image, but the highs too. An over-compressed program sounds pretty muddy to me. In the quest to get the level, they end up EQing the heck out of these tracks, which of course induces even more distortion between the EQ and the compression. I am hearing things that are very, very loud, but in my opinion not a very good sound. I am hearing a program that is just way over-EQed because they're trying to get back what the compressor has taken away.

How do you determine what's going to work and what isn't?
EDDY SCHREYER: By listening. You go as loud as you can, and you begin listening for digital clipping, analog grittiness, and things that begin to happen as you start to exceed the thresholds of what that mix will allow you to do, in terms of level. Again, just spanking as much gain as you can, be it in the analog or digital world, doesn't matter. You go for the level and properly control it with compression, then you start to EQ to achieve this balance. Of course, it all depends on the type of mix, how it was mixed, the kind of equipment that was used, how many tracks, the number of instruments, and the arrangement. Just the number of instruments can be a very limiting factor on level also. For example, a 96-track mix may not go as loud as a 24-track mix because there is too much signal to be processed.

You don't seem to compress things a lot, a dB and a half at the most. Is that typical?
EDDY SCHREYER: It's very typical of what I do with all my stuff, but I compress more than people are aware. I can compress in different stages, so hopefully you are not even really hearing it. You are not actually seeing the compression, either analog or digital, that I'm doing. But I do go a little lighter than a lot of other mastering houses.

Do you use multiple stages of compression then?
EDDY SCHREYER: Yes. I do use analog and digital compression and sometimes digital limiting. Sometimes I digitally limit, I digitally compress, and I analog compress. Very rarely do I use analog limiting, though. I use whatever is needed to control the program. In other words, when a program is mixed a little heavy on the snare, for example, I can use a digital limiter that will sort of clip the peak off that so that I can back off the dynamics of that particular instrument in the mix without EQing it out. Because if I go for the snare with EQ, I'm going to be pulling down the vocals and possibly the guitars as well. Likewise with the bass. If I go for a kick that's mixed too hot, adjusting 80, 60, 40 cycles or something to pull a kick down, it will really sacrifice the bottom quite a bit, so I'll tend to use

digital limiting to peak limit excessive dynamics in those particular cases. And then there's de-essing for sibilance on vocals and cymbals. That's all in trying to achieve balance again.

Do you think there is a difference in the way people master from city to city or coast to coast?
EDDY SCHREYER: Maybe slightly. And that only comes into play on the East Coast, for example. Certainly, I think there is competition on both coasts, but the East Coast might be a little more aggressive because of the competition between the mastering houses to be the king of the hill, so to speak.

So the sound is more aggressive.
EDDY SCHREYER: Absolutely. Whereas I think West Coast houses might be spread out a little more, so they are a little less aggressive with the style and type of mastering that's done. Which gets back primarily to level. It seems to me that the East Coast has gone a little overboard in the level game.

What do you think makes a great mastering engineer, as opposed to somebody who's just good?
EDDY SCHREYER: Probably the ability to hand-pick various pieces of equipment that maintain a sound. When I say maintain a sound, I mean keep the stereo separation strong. Also, the ability to use taste and know how far mastering can and can't go. Put it this way; a lot of times less is better.

Then you have the environmental issue. You can't make a move or create a fix if you can't hear it, so obviously the mastering environment is extremely important. Then, the ability to know just how far to push the creative envelope is important.

For example, I enjoy the creative editing possibilities when using the workstation in helping an album maintain some continuity and flow. If I hear something that will make a good crossfade, I'll mention it to the client. It may or may not fly, but we'll always try it. So I definitely like the creative part of the workstation, as it has created a great situation for mastering engineers to step forward and have a little more say in terms of the flow of the album with edits, spread times, and things like that. It's all part of the big picture, if you will, to keep the flow of an album happening.

What do you think makes for a great facility? And is it possible to have a great mastering engineer and a mediocre facility?
EDDY SCHREYER: A great facility to me means both client services and a comfortable place that's able to facilitate both large and small sessions. I am assuming my studio is somewhat the norm. I can seat about five to six

people in my room very comfortably and I believe that is probably somewhat common. I think a mastering room that's too small is not a good thing. At times there are more than two or three people who want to show up at a mastering session, so that part of the client relationship is very important to me. So the facility sort of dictates what your goal is in terms of the client/engineer relationship and just how comfortable you want these people to be. The client distractions are also one of the most important, yet simplest things—be it games or a nice kitchen where people can sit down and relax. Obviously staff is very important as well, in terms of helping clients, whether it be receiving a phone call or setting them up in a lounge to hear playback of various material. All of that, to me, represents a good facility.

Regarding the back end of that question, I've always felt, as a pretty good mastering engineer, that I've worked in some pretty lousy places. I'm one of those guys that might have been in lesser facilities until I got the chance to build my own. To some degree you can certainly have the ability and be hampered by budgetary concerns where equipment that you need is not being purchased. Or it could be just the physical limitations of the room, the size of the room, the type of monitors, or the sound of the room, which is certainly the most important thing. If the room is not there, I really believe you are in trouble. So some of the best guys have been locked down, I think, in lesser rooms.

Can you hear the final product in your head when you first do a run-through?
EDDY SCHREYER: Usually, yes I do. Typically, when I first put up a mix, the first thing I do is just go for the level without touching EQs unless there is something blatantly wrong. So I pretty much do get a picture in my head. The extreme is that a good mix is sometimes even more difficult to master in some respects than something that has a blatant problem, so I have got to be very careful because sometimes less is better.

Sometimes you throw up a mix and it's so kick-heavy with an 808, for example, that it is absolutely distorting from the get-go, so then you're tweaking right from the beginning. You immediately start to drop the bottom and try to get that balance going so you can dial out some of the kick, then the level starts increasing. I've mastered records where I pulled 4, 5, 6, 7, 8 dB out of the bottom, and all of a sudden I'm able to get 4 dB more overall program level. So when something is not balanced, it can really create big problems.

I do love the fact that vinyl is still hanging around because, ultimately, when a lot of these projects are cut to vinyl, that's what really susses engineers out. If they're distorting and mastered to the improper side of loud, it certainly doesn't go to vinyl well. Just the process of cutting vinyl is probably adding 15-percent distortion or more. The good news here at

Oasis is that we're hearing that our vinyl sounds better than anybody in the world at this point, and I'm very proud of that.

I know you cut vinyl for a long time, but you don't now. Do you miss it?
EDDY SCHREYER: Not terribly, no. It is a tedious process. I'm glad that I did cut vinyl, because again, that gets back to that big word "balance." The best sonic and the most properly mastered products always cut real well. The worst mastering jobs and the worst mixes master really badly. So I'm referring to this smoke-and-mirrors black art of balance, if you will, that's the toughest game, and cutting vinyl has probably been the biggest help in my entire career. Trying to get the audio balanced so that it would cut well was a huge help because a bad mastering job would cut just horribly. As you started balancing projects out properly, they would cut that much better.

Unfortunately, you can probably count the lacquer houses on one hand now in this country, so the new generation of mastering engineers has not had that training. As a result it's a little tougher to get to that final stage of mastering something well. Just like anything else, you can't have too much experience. I'm still learning every day because mastering is a constant learning experience. That's the good news, frustratingly so. The vinyl is just totally unforgiving, whereas the digital medium allows you to slam anything into it that you want, clipped or not, because it's not going to skip. In other words, you can almost do anything to a CD and get away with it. Left-right balance can be totally wrong, image can be totally wrong; it just doesn't matter because that CD will not skip. So basically, the taste factor becomes the limiting issue.

What's the hardest thing that you have to do? Do you get projects that are more difficult because of the way they're prepared or treated?
EDDY SCHREYER: I'd say one of the most difficult types of project is the one with source mismatches where some of it's on a file and some is on 1/2". I still find 1/2", properly aligned on good tape and a good machine, to be a deeper, wider sound. And I still enjoy listening to analog more than I do a lot of the files. But cutting an album with source mismatches is quite difficult because some of the digital formats sonically shrink to me. No matter what I do, that file is just going to sound a little thinner and a little less deep than the 1/2", so trying to create and maintain an album with flow and continuity in terms of sonics becomes difficult.

Soundtrack albums are probably the single most difficult type of project for me to do, especially if a score is involved. Sequencing is terribly important if score is coming behind a big rockin' song. It's very difficult because the score is dynamically wide with levels from maybe –20 to +3. The low-level score is never loud enough. I think it's always best to help

maintain good continuity and flow with good song sequencing. So maintaining some sort of sonic equality, if you will, on a soundtrack album is very difficult, especially if you're sequencing material that's maybe 10 or 15 years old and then current stuff. So probably the most difficult stuff outside of mismatching of sources would be the soundtrack album, but I enjoy doing them and I think I do them pretty well.

What makes your job easier? Is there something that a client can do to make everything go faster or smoother?
EDDY SCHREYER: Having some common sense, like being organized and obviously having a sequence in mind, helps. In general I'd always prefer to have the best mixes first. But if several studios were used for mixdown, I rather keep all the mixes from each studio together. So, if four or five different studios were used, I would start with all the tracks from studio number one. I don't care if it's song number 1, 3, 10, or 12; I would rather master those as a unit, and then move on to the next studio to keep some sort of continuity.

What's the thing that you enjoy most about mastering?
EDDY SCHREYER: The thing I enjoy most is taking a project to another level. And obviously, it's the greatest feeling in the world when Fiona Apple or Christina Aguilera or Offspring ends up being really outstanding sonically and then also achieves the sales that they do. It makes everybody involved with the project pretty happy.

Do you do all of your equalization, compression, and limiting before you hit the workstation?
EDDY SCHREYER: If the source is analog, it's the best of all worlds because then you're making just one digital conversion into the workstation, so that's the ultimate. I think it's silly to make an A-to-D conversion, process digitally, and then go back into the workstation. The less signal jacking the better, in my opinion.

I've noticed that you use a lot of little bits of EQ. Is that typical of most mastering guys?
EDDY SCHREYER: To tell you the truth, I don't really know how a lot of guys master their projects. I would suspect that I'm somewhat similar to a lot of guys, though. I tend to build sound versus stabbing things pretty strongly in one spot. That's about the easiest way as I can say it. I have digital and analog EQ, and upon listening the decision is made which should receive the bulk of the work.

How did you come by that method?
EDDY SCHREYER: Probably from tuning rooms using third-octave EQs. I tend to shape the sound, rather than stab it pretty strongly in spots.

How often do you have to add effects?
EDDY SCHREYER: Very rarely. I mean, it might happen twice a year in this room. We don't tend to get those sorts of problems.

Do you get people who pre-master things where they'll maybe cut intros off or cut fades off or something like that?
EDDY SCHREYER: Yes, sometimes for the worse. Usually they think they are saving time, but they might create more problems than if they left it alone in the first place. I've had some projects where they clipped intros and I've had to grab beats from other places and put them on the top, so I prefer it if you don't cut the program too tight. If there is a lot of very deliberate editing to be done and you want to save time and money offsite, then I understand it. But it better be right.

How important is mono to you? Do you listen in mono a lot?
EDDY SCHREYER: No, but I believe MTV uses a fold-in process, so there is certainly a consideration to be made for that. Depending on the mix, it's possible that certain instruments will disappear on the fold-in. So pure mono is really not a consideration at all, but if you're thinking of MTV at all, it is definitely a good idea to maybe narrow the spread just to maintain a little better match between a slight fold-in and pure stereo.

How did you go about choosing your monitors?
EDDY SCHREYER: I've been using Tannoys since about 1984 or '85. I'm just a big fan of the dual-concentrics. I think the phase coherency is just unsurpassed. Once you get used to listening to these boxes, it's very difficult to listen to spread drivers again. In this particular case, my Dual 15s have been custom-modified for the room to some degree, and using them is just a great treat. I think they are one of the easier speakers to listen to since they certainly don't sound like the big, brash monitor that they possibly might look to be. A typical comment made about the monitors here at Oasis is that they sound like the best big stereo system they've ever heard, which is a terrifically flattering compliment. I also have some little Tannoy System 600s for near-fields, and now I've added some dual 15 subs to the mains. Sonically speaking, I have been in quite a few rooms and I have yet to hear a system that rivals this, so I am very happy with it.

Tell me about the subwoofers. What was the reason for getting them, and why did you get two as opposed to one?
EDDY SCHREYER: My mains, the Dual 15s, are definitely light from, say, 30 Hz down, so I wanted to fill in the extreme lows more accurately because of the amount of R&B that I do. Darren Cavanaugh and Aria came up with a design that I just absolutely love. I feel I have a little bit better control with the pair than with a single sub in terms of where they sit. With one, you are pretty much locked down positionally, but with the two you actually have a little more flexibility.

Now that you have had some experience with surround sound, how do you feel about that as opposed to stereo?
EDDY SCHREYER: Oh, I am loving it, but it's a difficult medium to work in. It's not something you just throw up and do. To some degree you'd think it would be easier because you have five speakers to fill up instead of cramming all this information in two speakers, but it is not. The balance of the monitor system is extremely important, and the adjustment of levels of the drivers and then interfacing the sub is extremely critical on the mix. I find that the stereo image between the left and right, left and left surround, right and right surround in the crisscross from the left to the right surround is very, very tricky. I do hear some unusual low-frequency phase characteristics that I'm not real happy with, depending on the mix. I've also heard some very, very good mixes, so it can definitely work. But it is a difficult medium at best to really make sound good, but so is a really great stereo mix. 5.1 is just so new to all of us that it's much more difficult at this point, but when something is nailed, it's just awesome.

What's your favorite piece of gear?
EDDY SCHREYER: That is tough because the digital Weiss desk that I have certainly is still unsurpassed at this point. The Manley LimCom [Vari-Mu compressor] is definitely one of the best units I have in terms of analog. I really don't have a piece of gear in here that I dislike, so between Tube-Tech and Manley and Avalon, Waves L-2 and Junger, it is all my favorite stuff, to be honest with you. Sonically, it just doesn't let me down.

When you get handed a project, what are the steps? What do you actually go through on a whole project? Describe a whole project like Christina Aguilera, for example.
EDDY SCHREYER: Christina is an extreme example because of the complexity of the album. In other words, that particular album was mastered over the course of six to eight weeks, maybe longer. Songs were being remixed and getting swapped, so it was a little longer process than normal. Not that it was bad because, if anything, I didn't have to deal with the typical 12 or 13 songs in one day and nail them all with one mastering session. An average album rolls in where I am doing that in five to six hours, though.

Basically, a project starts out whereby a client comes in, hands me tapes, and gives me a song sequence. I just take it song by song and dump it into the workstation [an AudioCube] and then offload refs. The procedure can be relatively simple, outside of interludes and any special little musical pieces that may interface with the album in terms of spreads in between songs.

But Christina was unusual, as I say, because it was done over quite a period of time. That was actually great because as the sequence changed and songs came and went, my perspective on the sound of the album remained consistent because I was always given the time I needed.

Is it harder for you to do something like that over the course of a week or two than it is to do it all at once?
EDDY SCHREYER: It really depends. Sometimes I would say yes, but sometimes it gets crushingly difficult when a project just strings on and on and on because you can lose a bit of your objectivity.

I truly find that the R&B-type pop records are a little easier than rock records. Rock records get a little trickier because the balances are so critical. It just seems that a well-arranged R&B pop track is pretty simple for me to hear, whereas rock seems to need more sonic continuity than R&B tracks. It just feels better when they are seemingly coming from a similar place. Whereas R&B pop records can have much more extremes involved and it just plays out fine.

How does Latin stack up?
EDDY SCHREYER: It's similar. The only catch becomes—just as in my Japanese projects—that it's a little trickier to dissect vocal balances if they are not sung in English. I'll often turn to a client and ask about a word in Spanish or Japanese. "Was that okay? Was that discernable?" Because the Japanese market tends to go for a little higher vocal level because it is tough to hear the lyrics in the language. Ultimately, though, balance is still the key.

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Glossary

5.1 A speaker system that uses three speakers across the front and two stereo speakers in the rear, along with a subwoofer.

1630 A first-generation two-track digital tape machine utilizing a separate digital processor and a 3/4" U-matic video tape machine for storage. In the early years of the CD, 1630's were the primary master tape delivered to the pressing plant, but they are considered obsolete today. A model 1610 predated this machine.

acetate A single-sided vinyl check disc, sometimes called a *ref.* Due to the extreme softness of the vinyl, an acetate has a limited number of plays (five or six) before it wears out. (See *ref.*)

A/D Analog-to-digital converter. This device converts the analog waveform into the digital form of digital 1s and 0s.

AIFF Audio Interchange File Format (also known as *Apple Interchange File Format*) is the most used audio file format in the Apple Macintosh operating system. An AIFF file contains the raw audio data, channel information (monophonic or stereophonic), bit depth, sample rate, and application-specific data areas. The application-specific data areas let different applications add information to the file header that remains there even if the file is opened and processed by another application. For example, a file could retain information about selected regions of the audio data used for recalling zoom levels not used by other applications.

ambience The background noise of an environment.

asset A multimedia element—sound, picture, graphic, or text.

attack The first part of a sound. On a compressor/limiter, a control that affects how that device will respond to the attack of a sound.

attenuation A decrease in level.

Augsburger George Augsburger of Perception Inc. in Los Angeles is one of the most revered studio designers. He also designs large studio monitors, each having dual 15" woofers and a horn tweeter.

automation A system that memorizes then plays back the position of all faders and mutes on a console.

bandwidth The number of frequencies that a device will pass before the signal degrades. A human being can supposedly hear from 20 Hz to 20 kHz, so the bandwidth of the human ear is 20 Hz to 20 kHz. Sometimes applies to computer data rate, where a high rate per second represents a wider bandwidth.

bass management A circuit that utilizes the subwoofer in a 5.1 system to provide bass extension for the five main speakers. The bass manager steers all frequencies below 80 Hz into the subwoofer along with the LFE (see *LFE*) source signal.

bass redirection Another term for bass management.

bit rate The transmission rate of a digital system that is expressed in either kilobits per second (kbps) or megabits per second (Mbps).

bit splitter In order to record a signal with a 20- or 24-bit word length onto a recorder that is only 16-bit, the digital word is "split" across two tracks instead of one. This is sometimes known as *multiplexing*.

BLER Block Error Rate. A measurement of how many errors a disc contains. A BLER rate of 220 per second or above will cause the disc to be rejected, although the acceptable rate is usually far lower.

brick-wall A limiter employing "look-ahead" technology that is so efficient that no matter what happens, the signal will not exceed a certain predetermined level and there will be no digital "overs."

buss A signal pathway.

chamber (reverb) A method of creating artificial reverberation by sending a signal to a speaker in a tiled room that is picked up by several microphones placed in the room.

chorus A type of signal processor where a detuned copy is mixed with the original signal, which creates a fatter sound.

clipping When an audio signal begins to distort because of a circuit in the signal path being overloaded, the top of the waveform becomes "clipped" off and begins to look square instead of rounded. This usually results in some type of distortion, which can be either soft and barely noticeable or horribly crunchy sounding.

clone A copy of a tape that is bit-for-bit accurate with the original source.

codec Compressor/decompressor. A codec is a software algorithm that encodes and decodes a particular data format. Some examples of codecs are .mp3, .ac3, .wmv, and .flac.

comb filter A distortion produced by combining an electronic or acoustic signal with a delayed copy of itself. The result is peaks and dips introduced into the frequency response. This is what happens when a signal is flanged. (See *flanging.*)

competitive level A mix level that is as loud as your competitor's mix.

cut To decrease, attenuate, or make less.

cutter head The assembly on a lathe that holds the cutting stylus between a set of drive coils powered by very high-powered (typically 1,000- to 3,500-watt) amplifiers.

D/A Digital-to-analog converter. This device converts the digital 1s and 0s back to an analog waveform.

DAT Digital Audio Tape. An inexpensive digital audio format using 4mm-wide tape. This format was originally intended for the consumer market, but has found widespread use in professional circles due to its small size and low cost.

data compression Data compression is the process of using psychoacoustic principles to reduce the number of bits required to represent the signal.

DAW Digital Audio Workstation. A computer with the appropriate hardware and software needed to digitize and edit audio.

DDP Disc Description Protocol. A proprietary format developed by Doug Carson Associates that is low in errors and allows high-speed glass master cutting. It is currently the standard delivery format for CDs and DVDs.

decay The time it takes for a signal to fall below audibility.

delay A type of signal processor that produces distinct repeats (echoes) of a signal.

digital domain When a signal source is digitized, or converted into a series of electronic pulses represented by 1s and 0s, the signal is then in the digital domain.

dipole A loudspeaker having a figure-eight directional pattern and often used for reproducing the surround channels of a multichannel audio system by placing the listening area in the null of the figure-eight pattern. Dipoles are often found to be better at reproducing enveloping sounds, such as reverberation and ambience, and poorer at localizing than a direct radiator. Also, dipoles simulate an array of loudspeakers in theaters when used in the home.

direct radiator A loudspeaker where the principal output is directed at the listening area. Universally used for the front channels in a multichannel sound system and widely used for the surround channels, direct radiators are often found to be better for localization and poorer for diffuse-field reproduction, such as for reverberation and ambience, than dipole radiators.

distressor A compressor made by Empirical Labs that's noted for its distinctively aggressive sound.

dither A low-level noise signal used to reduce the distortion that sometimes occurs when reducing the length of a digital word.

DLT Digital Linear Tape. A high-speed, large-capacity format for data backup. Also used as the standard master for DVD delivery to the replicator.

Dolby Digital A data compression method, otherwise known as *AC-3*, that uses psychoacoustic principles to reduce the number of bits required to represent the signal. Bit rates for 5.1 channels range from 320 kbps for sound on film to 384 kbps for digital television and up to 448 kbps for audio use on DVD. AC-3 is also what is known as a *lossy* compressor (see *lossy compression*), which relies on psychoacoustic modeling of frequency and temporal masking effects to reduce bits by eliminating those parts of the signal thought to be inaudible. The bit-rate reduction achieved at a nominal 384 kbps is about 10:1.

Dolby Pro Logic An active matrix decoder that extracts four signals from two-channel Dolby Surround–encoded material. The four channels are left, center, and right front channels, and a single-bandwidth limited mono surround channel. The amplitude-phase matrix decoder uses level difference between the two source channels, called Lt and Rt, to steer across left-center-right, and the phase difference to steer from front to surround.

Dolby Surround A digital encoding system that combines four channels (left, center, right, and a limited-bandwidth surround channel) into two channels. These two channels can be summed together for mono playback or played back as normal stereo. When the two channels are fed into the active Dolby Pro Logic decoder, the matrix is unfolded back into four channels again. The limited-bandwidth surround channel is reproduced through the left surround and right surround speakers. If the matrix is fed into a passive decoder, then only the stereo signal plus the surround channel is unfolded.

downmix To automatically extract a stereo or mono mix from an encoded surround mix.

DSP Digital Signal Processing. Processing within the digital domain, usually by dedicated microprocessors.

DTS Digital Surround A data-compression method developed by Digital Theater Systems using waveform coding techniques that takes six channels of audio (5.1) and folds them into a single digital bit stream. This differs from Dolby Digital in that the data rate is a somewhat higher 1.4 Mbps, which represents a compression ratio of about 4:1. DTS is also what's known as a *lossy* compression. (See *lossy compression.*)

DTV Digital Television.

dynamic range A ratio that describes the difference between the loudest and the quietest audio. The higher the number, the better.

element A component or ingredient of the mix.

elliptical EQ A special equalizer built especially for vinyl disc mastering that takes excessive bass energy from either side of a stereo signal and directs it to the center. This prevents excessive low-frequency energy from cutting through the groove wall and destroying the master lacquer.

equalizer A tone control that can vary in sophistication from very simple to very complex. (See *parametric equalizer.*)

exciter An outboard effects device that uses phase manipulation and harmonic distortion to produce high-frequency enhancement of a signal.

feathering A technique used in applying EQ so that rather than applying a large amount of equalization at a single frequency, small amounts are added instead at the frequencies adjoining the one of principle concern.

flanging The process of mixing a copy of the signal back with itself, but gradually and randomly slowing the copy down to cause the sound to "whoosh" as if it were in a wind tunnel. This was originally done by holding a finger against a tape flange (the metal part that holds the tape on the reel), hence the name.

Fletcher-Munson curves A set of measurements that describes how the frequency response of the ear changes at different sound pressure levels. For instance, we generally hear very high and very low frequencies much better as the overall sound pressure level is increased.

glass master The first and most important step in CD replication, from which the stampers are eventually made.

groove The pulse of the song and how the instruments dynamically breathe with it. Or, the part of a vinyl record that contains the mechanical information that is transferred to electronic info by the stylus.

HDCD High Definition Compatible Digital. A process by Pacific Microsonics that encodes 20 bits of information onto a standard 16-bit CD while still remaining compatible with standard CD players.

headroom The amount of dynamic range between the normal operating level and the maximum output level, which is usually the onset of clipping.

high-pass filter An electronic frequency filter that allows only the high frequencies to pass. The frequency point where it cuts off is usually either switchable or variable.

hypercompression Too much buss compression during mixing or limiting during mastering in an effort to make the recording louder results in what's known as *hypercompression*, a condition that essentially leaves no dynamics and makes the track sound lifeless.

I/O The input/output of a device.

jitter The AES/EBU waveform should have particular transitions at precise intervals. Jitter is a measure of the instability of this timing. Timing errors result in frequency modulation of the audio signal, which, in extreme cases, can be detected as side bands on either side of a constant tone.

lacquer The vinyl master, which is a single-sided 14" disc made of aluminum substrate covered with a soft cellulose nitrate. A separate lacquer is required for each side of a record. Since the lacquer can never be played, a ref or acetate is made to check the disc. (See *ref* and *acetate.*)

LBR Laser Beam Recorder. The device that cuts the glass master from which the CD stampers are made.

LFE Low-frequency effects channel. This is a special channel of 5- to 120-Hz information primarily intended for special effects, such as explosions in movies. The LFE has an additional 10 dB of headroom in order to accommodate the required level.

look-ahead In a mastering limiter, look-ahead delays the audio signal a small amount (about two milliseconds or so) so that the limiter can anticipate the peaks in such a way that it catches the peak before it gets by.

lossless compression A compression format that recovers all the original data from the compressed version. MLP, Dolby TrueHD, and DTS-HD Master Audio are lossless compression schemes.

lossy compression A compression format that cannot recover all of its original data from the compressed version. Supposedly some of what is normally recorded before compression is imperceptible, with the louder sounds masking the softer ones. As a result, some data can be eliminated since it's not heard anyway. This selective approach, determined by extensive psychoacoustic research, is the basis for lossy compression. It is debatable, however, how much data can actually be thrown away (or compressed) without an audible sacrifice. Dolby Digital and DTS are lossy compression schemes.

low-pass filter A electronic frequency filter that allows only the low frequencies to pass. The frequency point where it cuts off is usually either switchable or variable.

LPCM Linear Pulse Code Modulation. This is the most common method of digital encoding of audio used today and is the same digital encoding method used by current audio CDs. In LPCM, the analog waveform is measured at discrete points in time and converted into a digital representation.

makeup gain A control on a compressor/limiter that applies additional gain to the signal. This is required since the signal is automatically decreased when the compressor is working. Makeup gain "makes up" the gain and brings it back to where it was prior to being compressed.

mastering The process of turning a collection of songs into a record by making them sound like they belong together in tone, volume, and timing (spacing between songs).

metadata Data that describes the primary data. For instance, metadata can be data about an audio file that indicates the date recorded, sample rate, resolution, and so on.

MLP Meridian Lossless Packing. A data-compression technique designed specifically for high-quality (96-kHz/24-bit) sonic data. MLP differs from other data-compression techniques in that no data is thrown away, thereby claiming the "lossless" moniker. MLP is also a standard for the 96-kHz/24-bit portion of the new DVD-Audio disc and will be licensed by Dolby Labs.

MO Magneto Optical. A writable method of digital storage utilizing an optical disc. Each disc stores from 250 MB to 4.3 GB and may be double-sided. Its widespread use has been limited due to its slow disc access time.

modulate The process of adding a control voltage to a signal source in order to change its character. For example, modulating a short slap delay with a 0.5-Hz sine wave will produce chorusing. (See *chorus.*)

mother In either vinyl or CD manufacturing, the intermediate step from which a stamper is made.

mute An on/off switch. To mute something means to turn it off.

noise shaping Dither that moves much of the injected noise to an audio band beyond what we can hear.

normalization A selection on a DAW that looks for the highest peak of an audio file and adjusts all the levels of the file upward to match that level.

overs Digital overs occur when the level is so high that it tries to go beyond 0 dB full scale on a typical digital level meter found in just about all equipment. A red overload indicator usually will turn on, accompanied by the crunchy, distorted sound of waveform clipping.

parametric equalizer A tone control in which the gain, frequency, and bandwidth are all variable.

parts The different masters sent to the pressing plant. A mastering house may make different parts/masters for CD, cassette, and vinyl or send additional parts to pressing plants around the world.

phantom image In a stereo system, if the signal is of equal strength in the left and right channels, the resultant sound appears to come from in between them. This is a phantom image.

phase shift The process during which some frequencies (usually those below 100 Hz) are slowed down ever so slightly as they pass through a device. This is usually exaggerated by excessive use of equalization and is highly undesirable.

pitch On a record, the velocity of the cutter head. Measured by the number of lines (grooves) per inch.

plate (reverb) A method to create artificial reverberation using a large steel plate with a speaker and several transducers connected to it.

PMCD Pre-Mastered CD. An obsolete format similar to a CD-R, except that it has PQ codes written on the lead out of the disc to expedite replication.

PQ codes Subcodes included along with the main data channel as a means of placing control data, such as start IDs and tables of contents, on a CD.

predelay A variable length of time before the onset of reverberation. Predelay is often used to separate the source from the reverberation so the source can be heard more clearly.

Pultec An equalizer sold during the ’50s and ’60s by Western Electric that is highly prized today for its smooth sound.

pumping When the level of a mix increases, and then decreases noticeably. Pumping is caused by the improper setting of the attack and release times on a compressor.

punchy A description for a quality of sound that infers good reproduction of dynamics with a strong impact. The term sometimes means emphasis in the 200-Hz and 5-kHz areas.

Q Bandwidth of a filter or equalizer.

range On a gate or expander, a control that adjusts the amount of attenuation that will occur to the signal when the the signal drops below the threshold.

ratio A parameter control on a compressor/limiter that determines how much compression or limiting will occur when the signal exceeds the threshold.

recall A system that memorizes the position of all pots and switches on a console. The engineer must still physically reset the pots and switches back to their previous positions as indicated on a video monitor.

Red Book The pre-recorded CD audio standard that you find in music stores today. Because of this standard, any CD will play in any audio compact disc player. Specified are the sample rate (44.1 kHz), bit depth (16), type of error detection and correction, and how the data is stored on the disc, among other things.

ref Short for *reference record*, a ref is a single-sided vinyl check disc, sometimes called an *acetate*. Due to the extreme softness of the vinyl, a ref has a limited number of plays (five or six) before it wears out. (See *acetate*.)

reference level This is the audio level, either electronic and acoustic, at which a sound system is aligned.

release The last part of a sound. On a compressor/limiter, a control that affects how that device will respond to the release of a sound.

return Inputs on a recording console especially dedicated for effects devices, such as reverbs and delays. The return inputs are usually not as sophisticated as normal channel inputs on a console.

reverb A type of signal processor that reproduces the spatial sound of an environment (for example, the sound of a closet or locker room or inside an oil tanker).

RIAA curve An equalization curve instituted by the Recording Industry Association of America (the RIAA) in 1953 that narrowed the grooves, thereby allowing more of them to be cut on a record, which increased the playing time and decreased the noise. This was accomplished by boosting the high frequencies by about 17 dB at 15 kHz and cutting the lows by 17 dB at 50 Hz when the record was cut. The opposite curve is then applied during playback.

sample rate The rate at which the analog waveform is measured. The more samples per second of the analog waveform that are taken, the better the digital representation of the waveform that occurs, resulting in greater bandwidth for the signal.

sampling An analog audio waveform is measured by an analog-to-digital converter (called an *A-to-D*, *ADC*, or *A/D converter*) in amplitude at discrete points in time and converted from electronic data to digital data.

scalability A feature of DVD-A that allows the producer to select from various sample rates (44.1, 48, 88.2, 96, 176.4, and 192 kHz) and word lengths (16, 20, 24). It is also possible for the producer to assign different sample rates and word lengths to different channel families, such as 96/24 to the front speakers and 48/16 to the surrounds.

SDDS Sony Dynamic Digital Sound. Sony's digital delivery system for the cinema. This 7.1 system features five speakers across the front, stereo speakers on the sides, plus a subwoofer.

sibilance A rise in the frequency response in a vocal where there's an excessive amount of upper midrange frequencies, resulting in the "S" sounds being overemphasized.

slate A comment added to a tape or track to identify it. In the early days of tape, a 50-Hz slate tone was added before each take of a song to easily identify its beginning as the tape was rewinding.

source tape An original master tape that is not a copy or a clone.

Spatializer A process developed by Spatializer Laboratories that uses psychoacoustic algorithms to give the listener the impression that he is immersed in sound.

SPL Sound Pressure Level.

SRC Sample Rate Conversion.

stamper In either vinyl or CD manufacturing, a negative copy bolted into the presser to actually stamp out records or CDs.

stems Mixes that have their major elements broken out separately for individual adjustment at a later time.

sub Short for subwoofer.

subwoofer A low-frequency speaker with a frequency response from about 25 Hz to 120 Hz.

synchronization When two devices—usually storage devices such as tape machines, DAWs, or sequencers—are locked together with respect to time.

test tones A set of tones used to calibrate a playback system. In the days of tape, they were added to a tape to help calibrate the playback machine.

threshold The point at which an effect takes place. On a compressor/limiter, for instance, the threshold control adjusts the point at which compression will take place.

THX A set of specifications, primarily for movie theaters, that specifies the acoustics and playback equipment so a movie will sound reasonably the same from theater to theater. THX was created by audio scientist and educator Tomlinson Holman for Lucasfilm. THX stands for *Tomlinson Holman Experiment.*

TV mix A mix without the vocals so the artist can sing live to the back tracks during a television appearance.

unity gain When the output level of a process or processor exactly matches its input level.

UDF Universal Disc Format. The file system used by DVD that eliminates much of the confusion that CD-ROM had due to the many different file formats used. All DVD formats use UDF and, as a result, have some level of compatibility with not only all DVD players, but also with computers using DOS, OS/2, Windows, Mac, and UNIX operating systems.

U-matic An industrial video machine utilizing a cassette storing 3/4" tape. The U-matic is the primary storage device for the 1630 digital processor.

variable pitch On a record, varying the number of grooves per inch depending upon the program material.

varispeed A parameter on tape recorders that varies the speed of the playback.

vinylite The vinyl used to make records actually comes in a granulated form called *vinylite.* Before being pressed, it is heated into the form of modeling clay and colored with pigment.

WAV A WAV file is an audio data file developed by the IBM and Microsoft corporations and is the PC equivalent of an AIFF file. It is identified by the .wav file extension.

word length The number of bits in a word. Word length is in groups of eight. The longer the word length, the better the dynamic range.

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